# Advanced Networks

Transport layer: TCP



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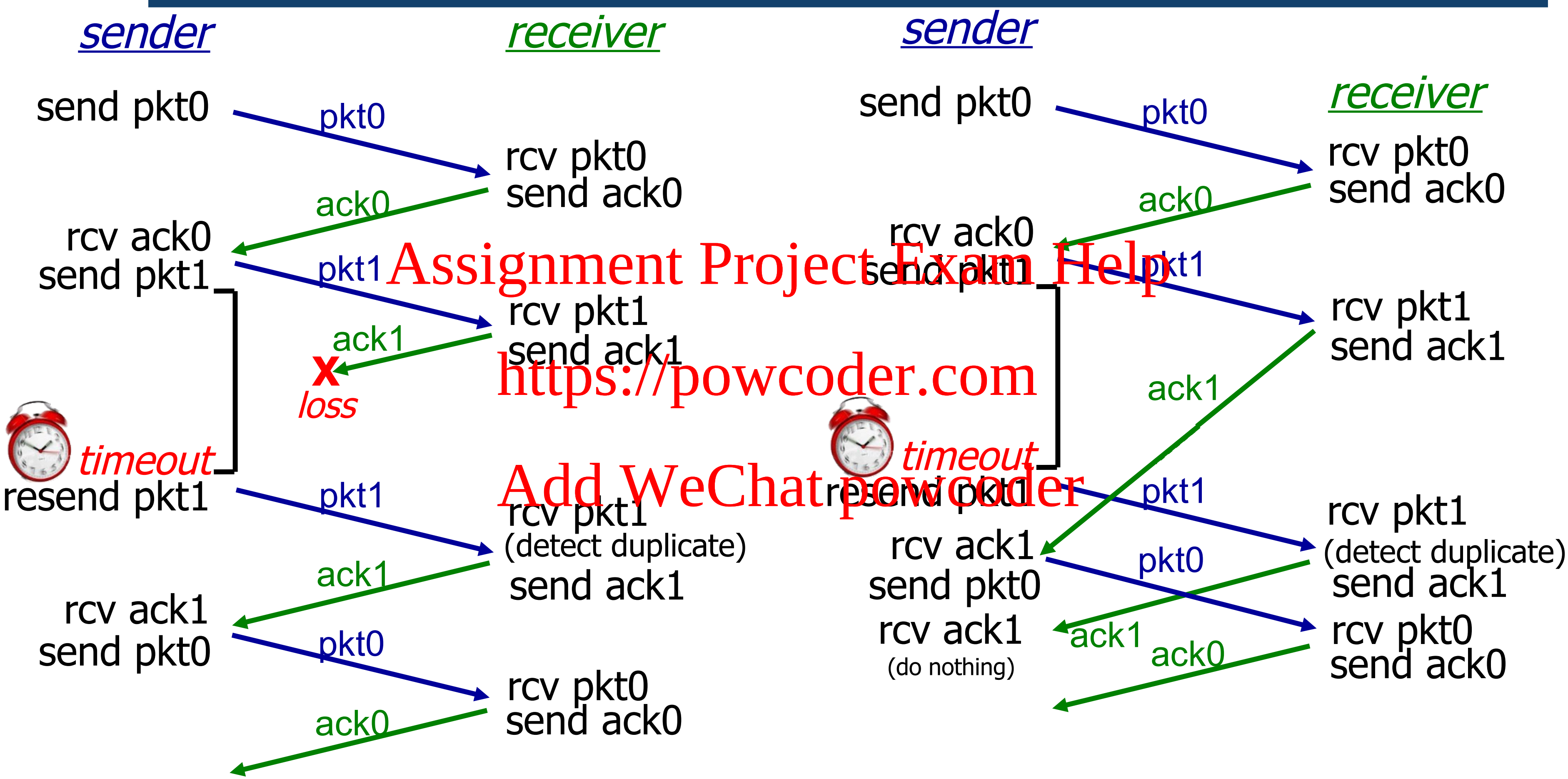
# rdt3.0 in action




(c) ACK loss

(d) premature timeout/ delayed ACK

# Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:



$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$



- $U_{sender}$: *utilization* – fraction of time sender busy sending

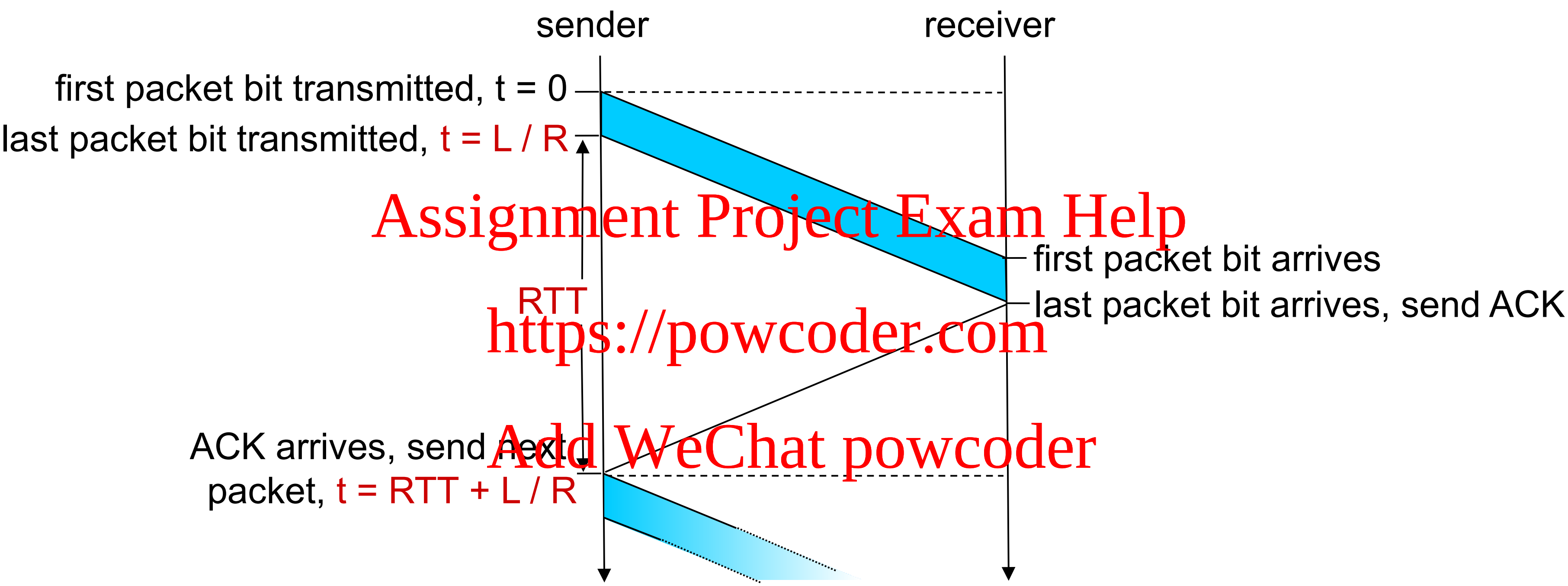


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

- if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link

- network protocol limits use of physical resources!

# rdt3.0: stop-and-wait operation





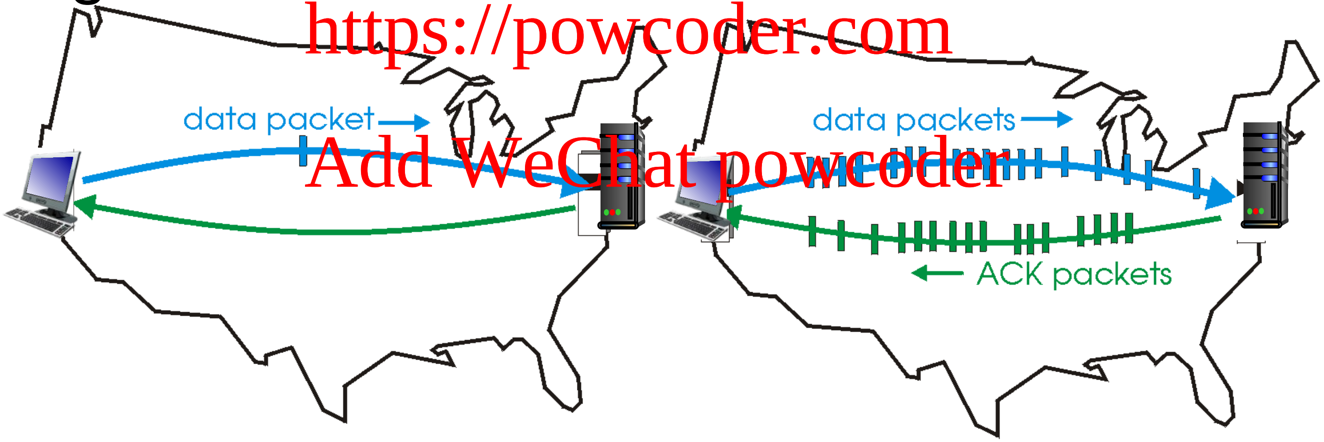

$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

# Pipelined protocols

pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

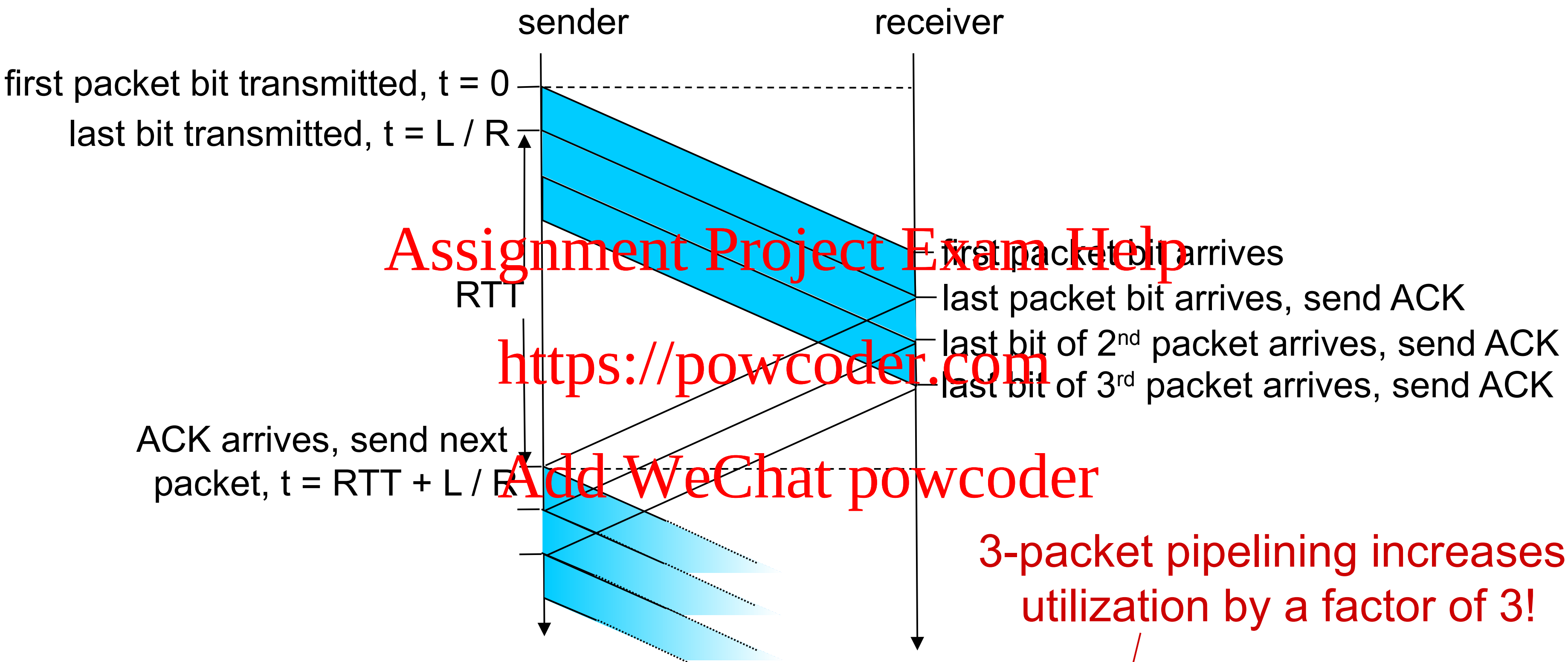





(a) a stop-and-wait protocol in operation (b) a pipelined protocol in operation

› two generic forms of pipelined protocols: *go-Back-N, selective repeat*

# Pipelining: increased utilization






3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

# Pipelined protocols: overview

## Go-back-N:

- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - does not ack packet if there is a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

## Selective Repeat:

- sender can have up to N unacked packets in pipeline
- receiver sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

# Go-Back-N: sender

› "window" of up to N, consecutive unacked pkts allowed




- ACK(n): ACKs all pkts up to, including seq # n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n)*: retransmit packet n and all higher seq # pkts in window

# Go-Back-N: sender

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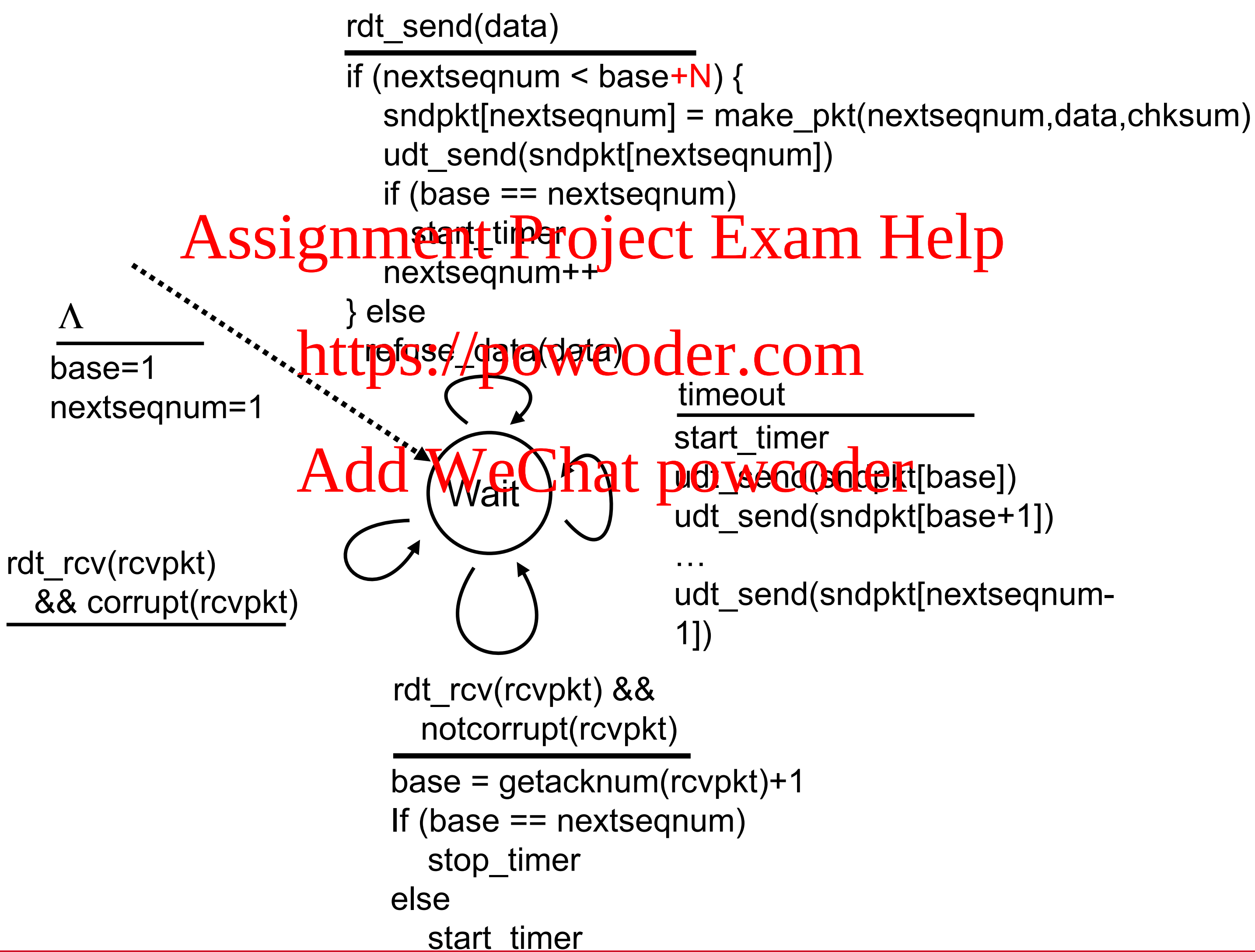
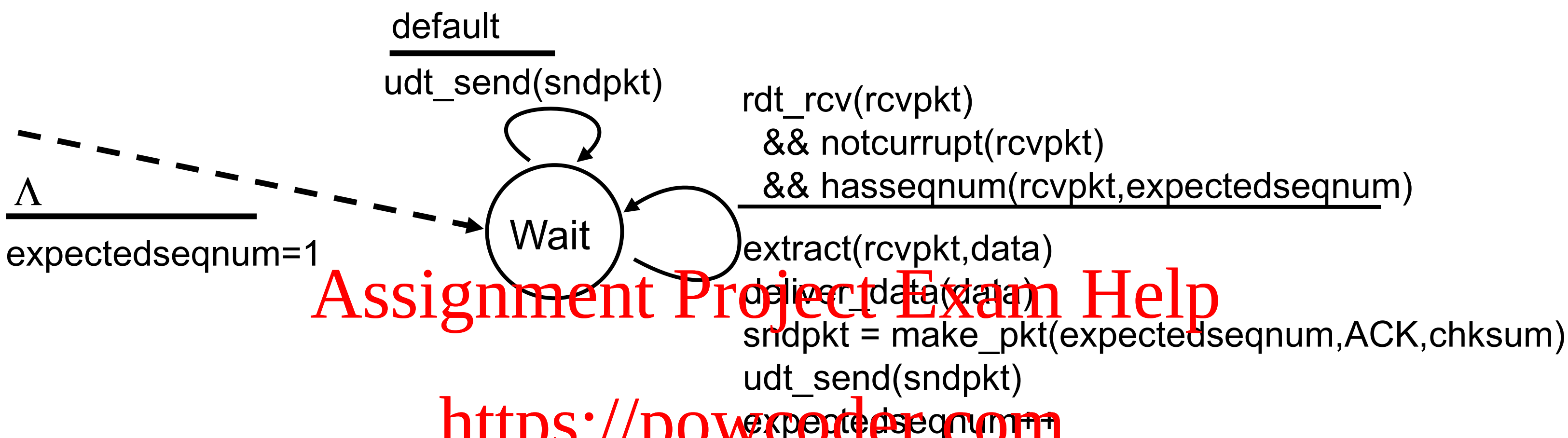
# GBN: sender extended FSM



# GBN: receiver extended FSM





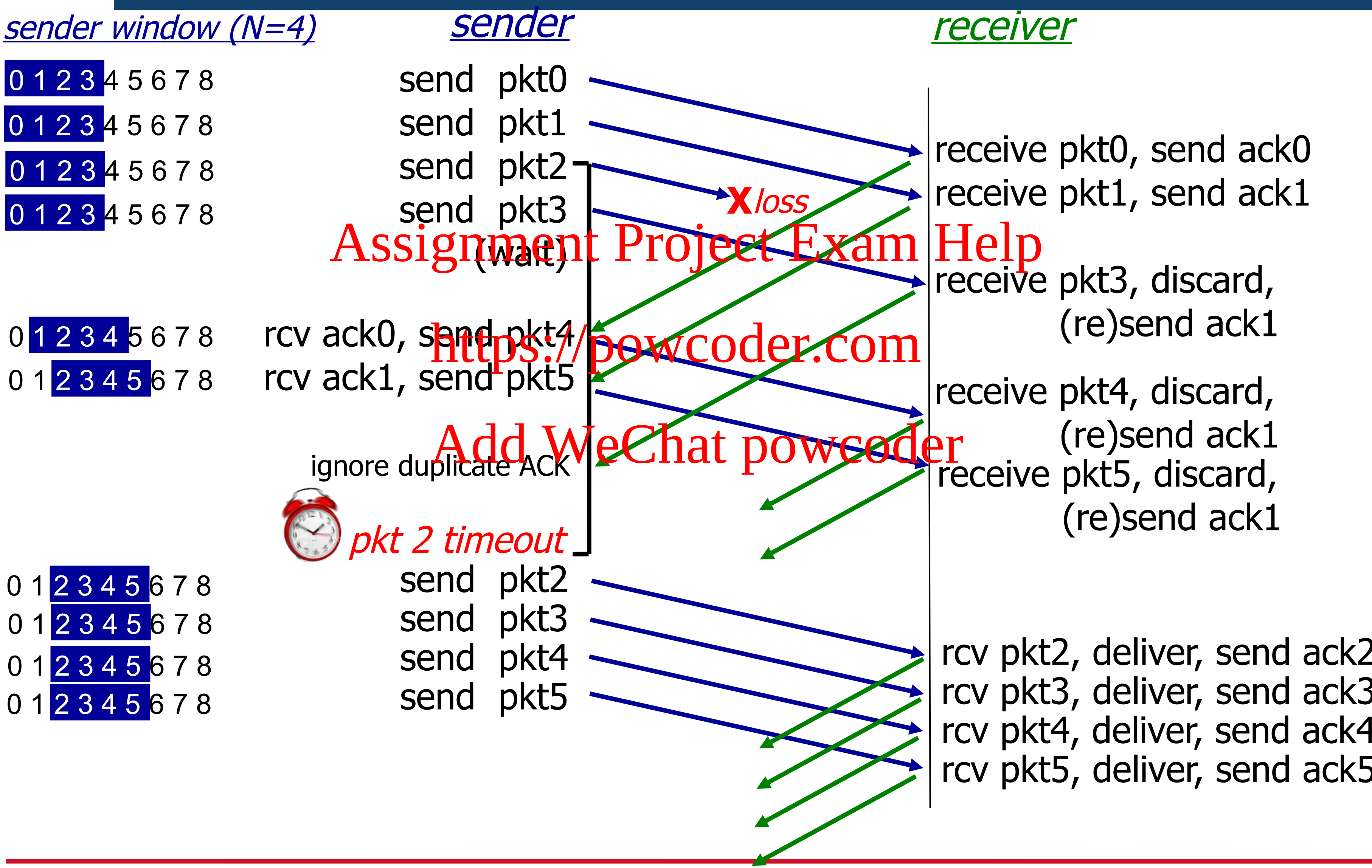

ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **expectedseqnum**

› out-of-order pkt:

- discard (don't buffer): *no receiver buffering!*
- re-ACK pkt with highest in-order seq #

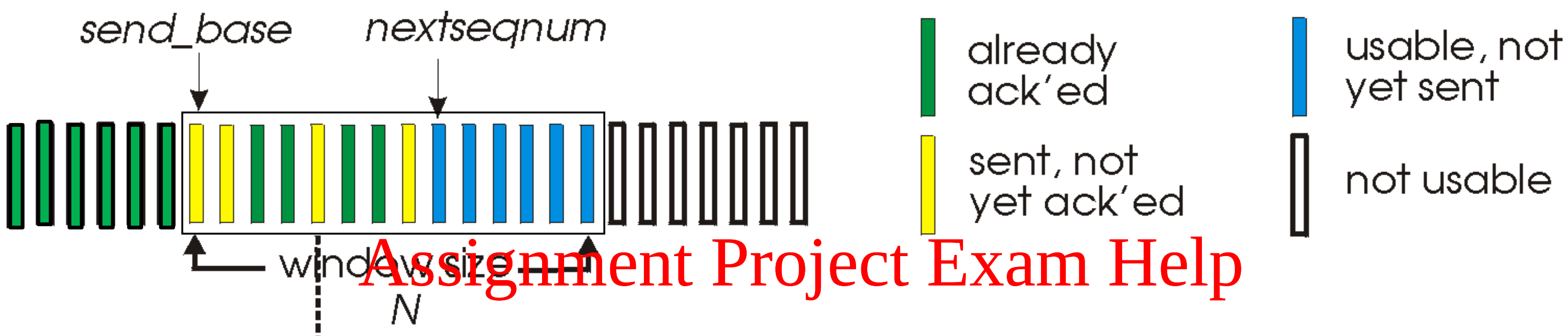
# GBN in action



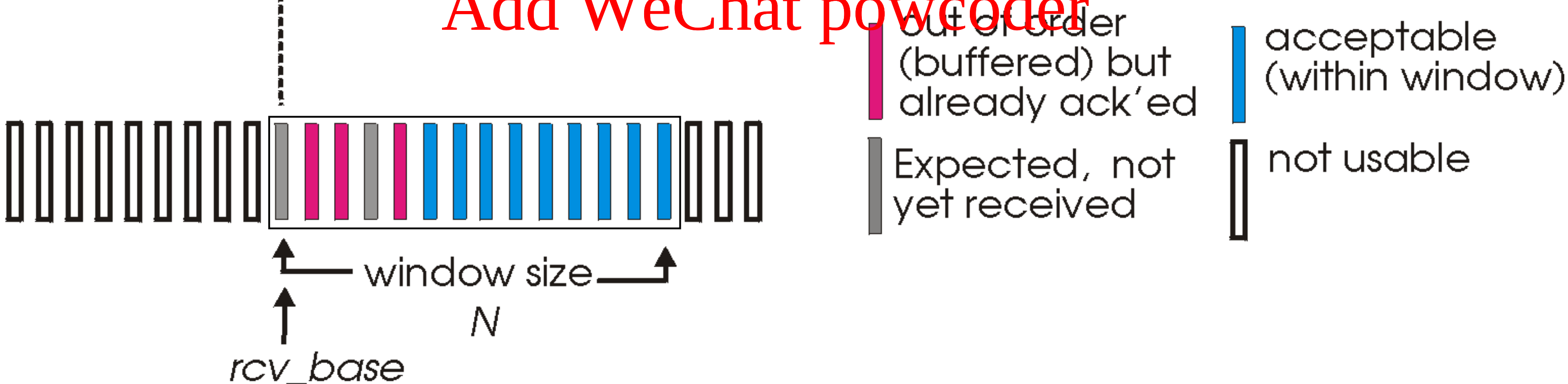
# Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
- receiver window

# Selective repeat: sender, receiver windows




(a) sender view of sequence numbers




(b) receiver view of sequence numbers

# Selective repeat

## sender

**data from above:**

› if next available seq # in window, send pkt

**timeout(n):**

› resend pkt n, restart timer

**ACK(n)** in [sendbase,sendbase+N-1]:

› mark pkt n as received

› if n is smallest unACKed pkt, advance window base to next unACKed seq #

## receiver

**pkt n in** [rcvbase, rcvbase+N-1]

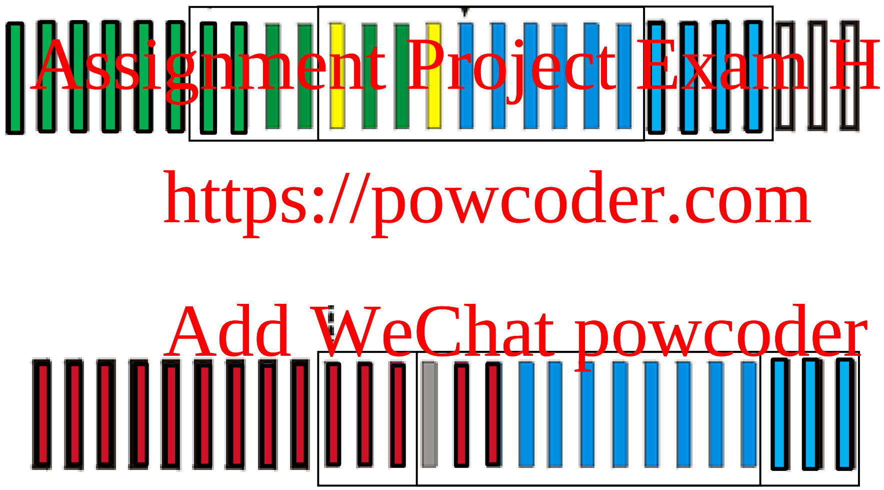
- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

**pkt n in** [rcvbase-N,rcvbase-1]

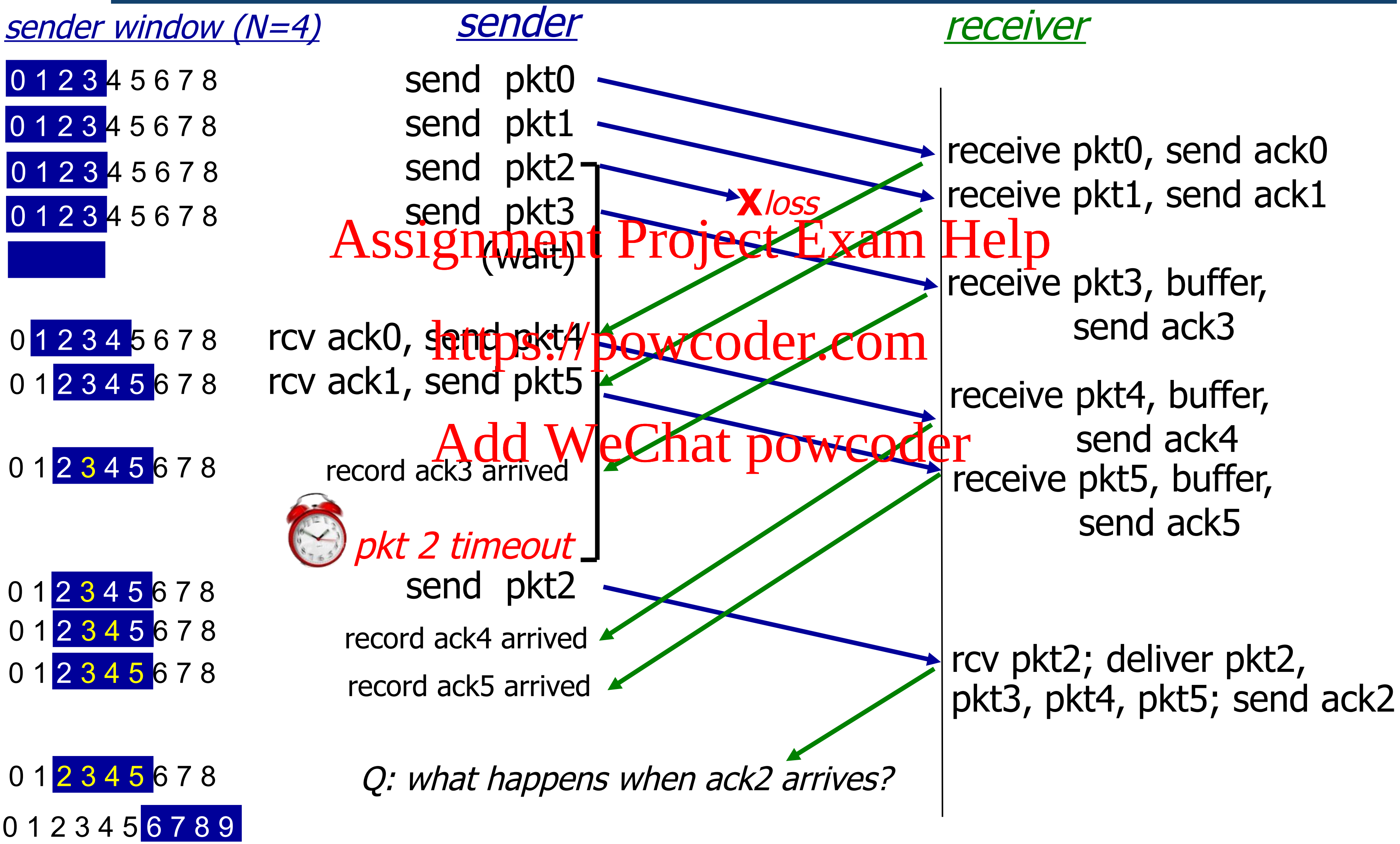
- ACK(n)

**otherwise:**

- ignore



# Selective repeat in action

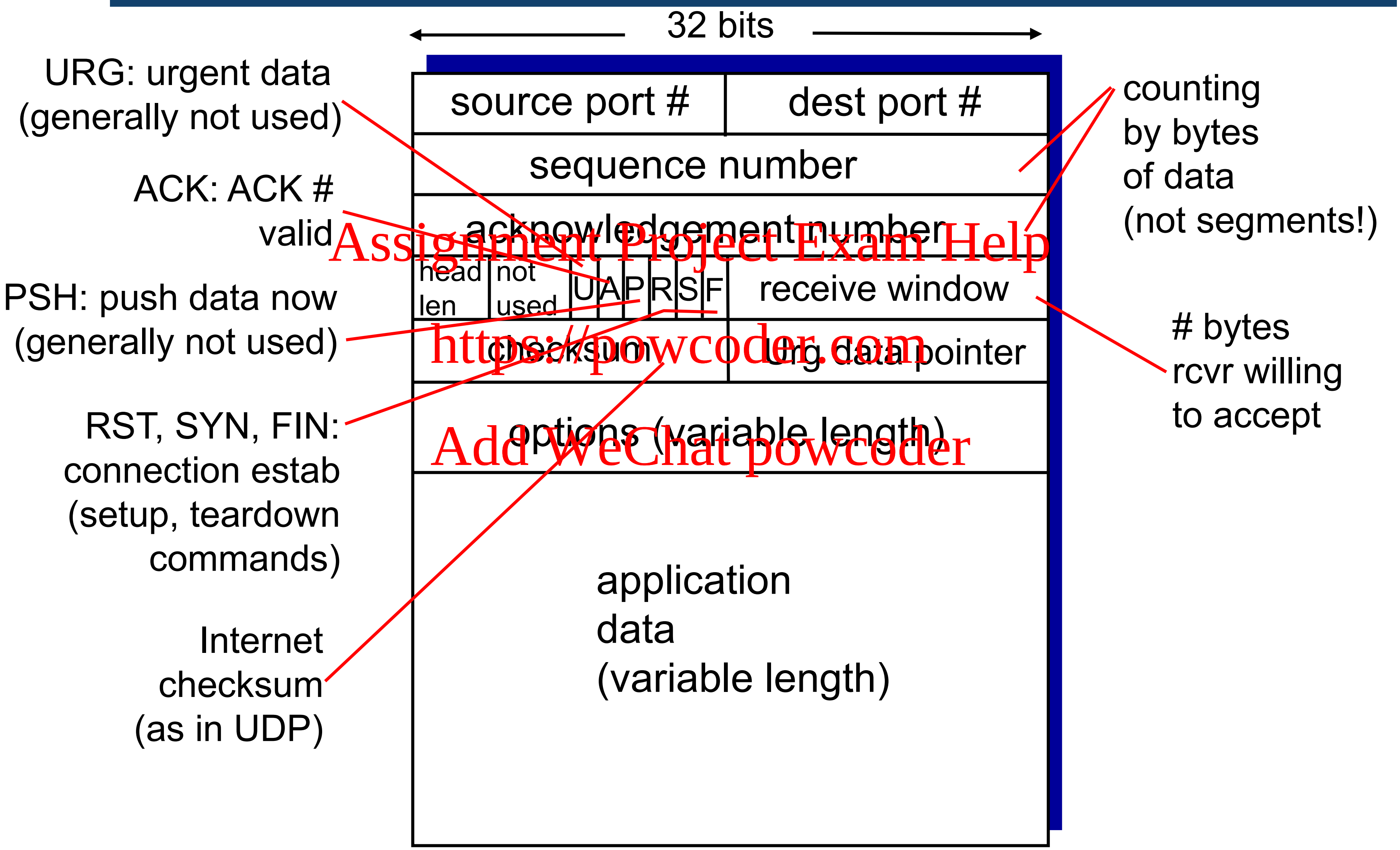


# Connection-oriented Transport TCP

# TCP: Overview RFCs: 793,1122,1323, 2018, 2581

› point-to-point:
  - one sender, one receiver
› reliable, in-order *byte stream*
› pipelined:
  - TCP congestion and flow control set window size
› full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
› connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
› flow controlled:
  - sender will not overwhelm receiver

# TCP segment structure



# TCP seq. numbers, ACKs

sequence numbers:

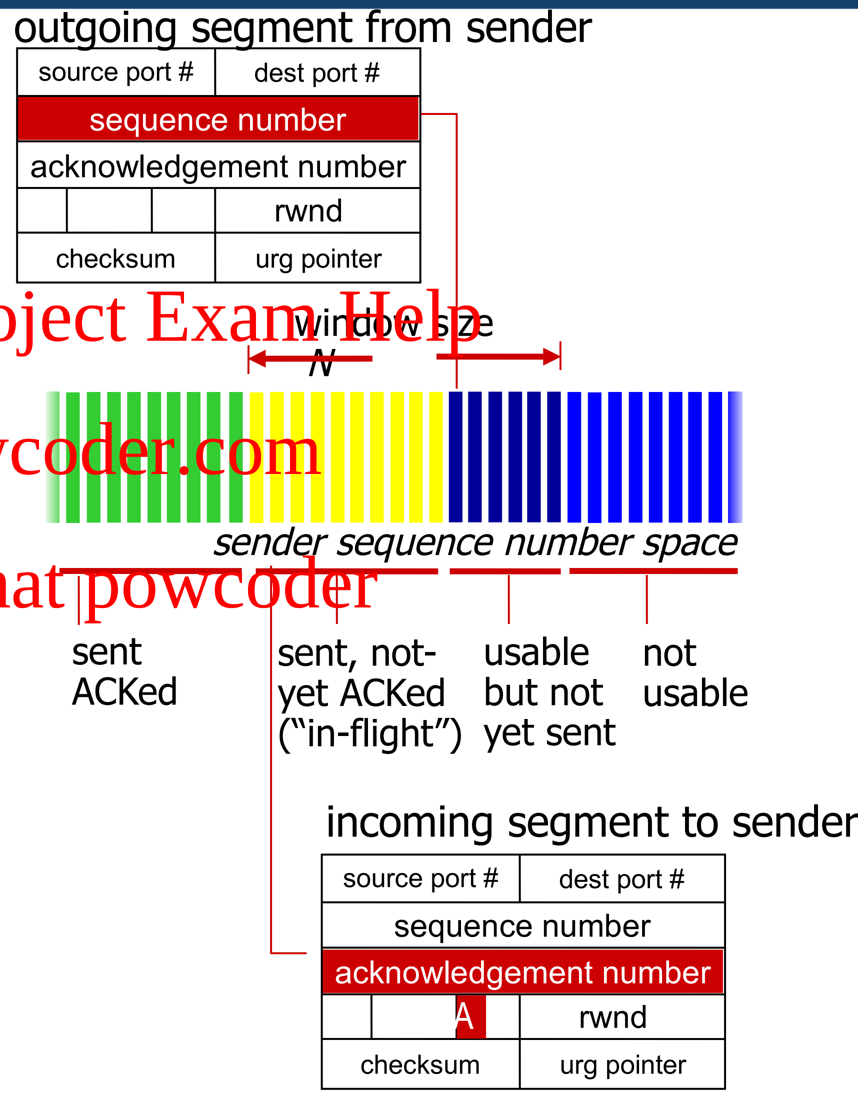
- "number" of first byte in segment's data

acknowledgements:

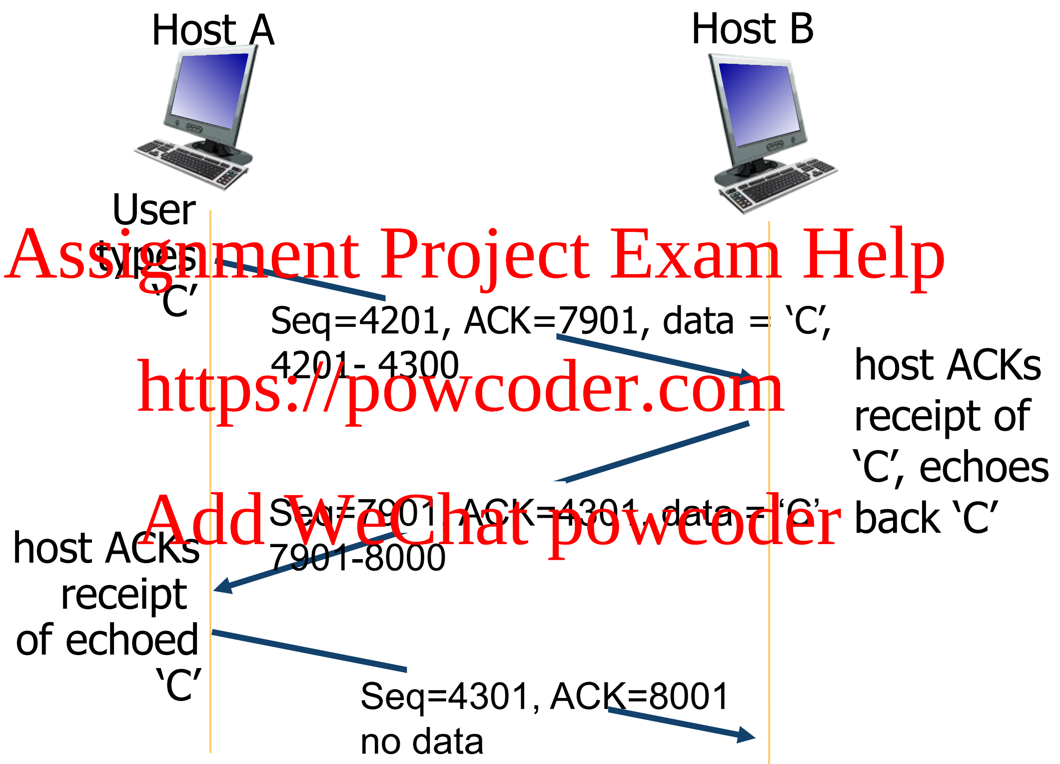
- seq # of **next byte** expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say,
- up to implementor
- Most will store, but still use cumulative ACK



# TCP seq. numbers, ACKs






simple telnet scenario

# TCP round trip time, timeout

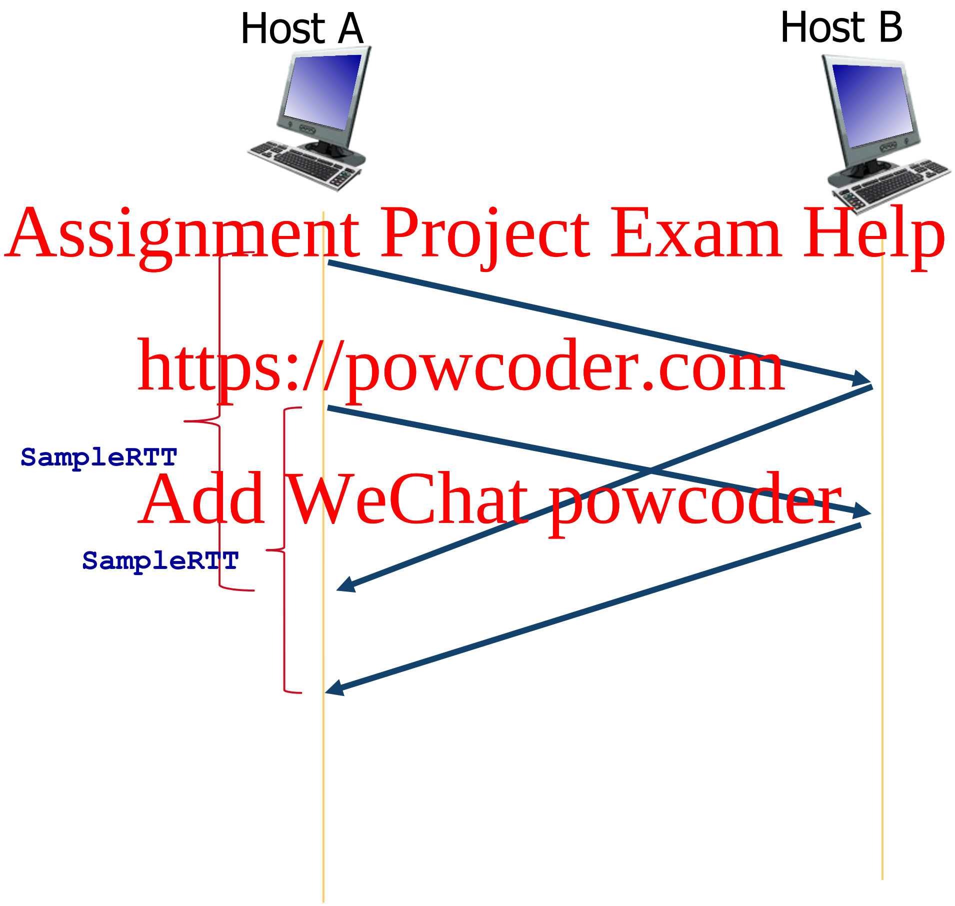
Q: how to set TCP timeout value?

- longer than RTT
  - but RTT varies
- *too short:* premature timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

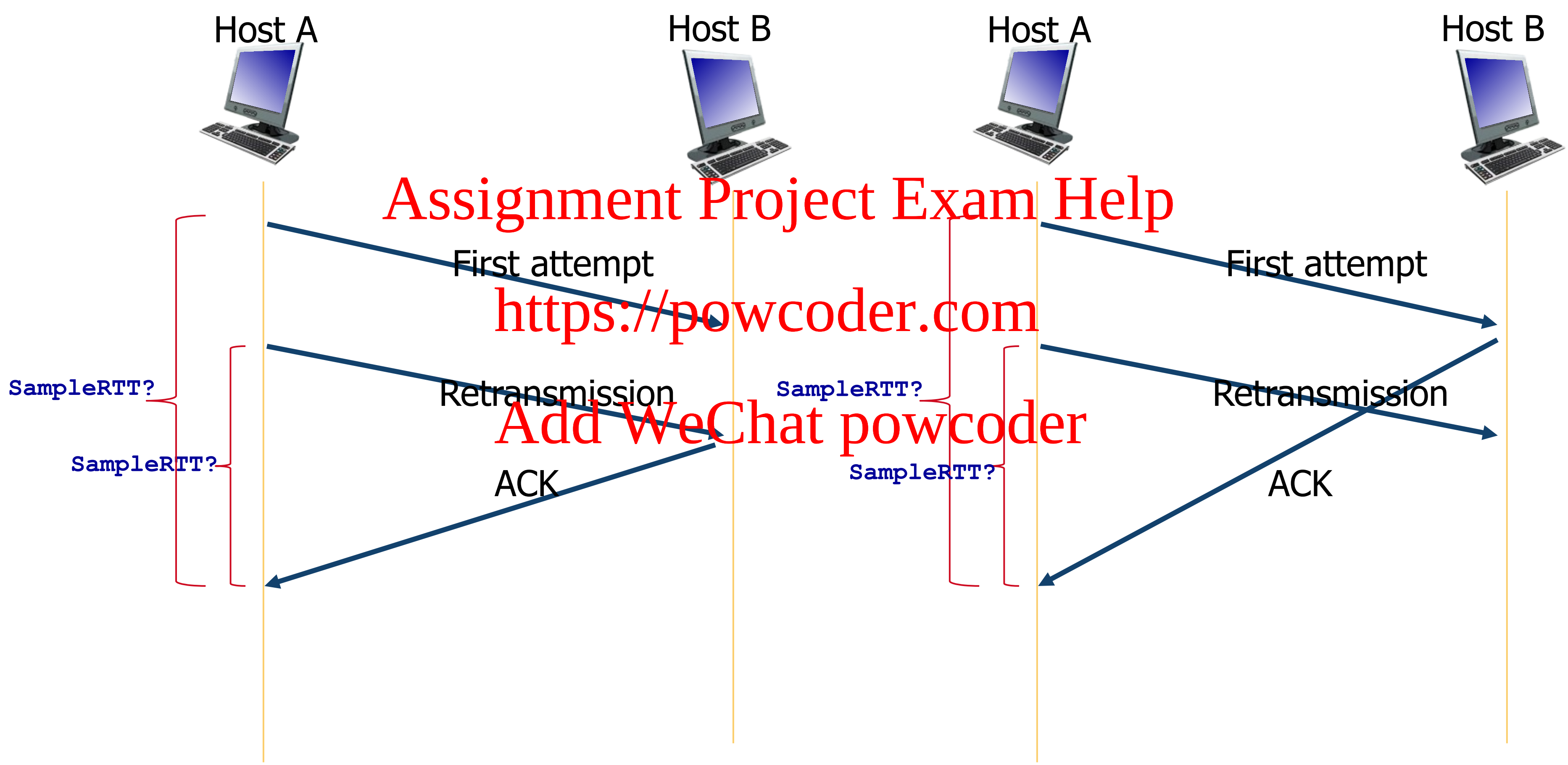
Q: how to estimate RTT?

- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - weighted average of several *recent* measurements, not just current **`SampleRTT`**

# SampleRT



# Ignore retransmissions

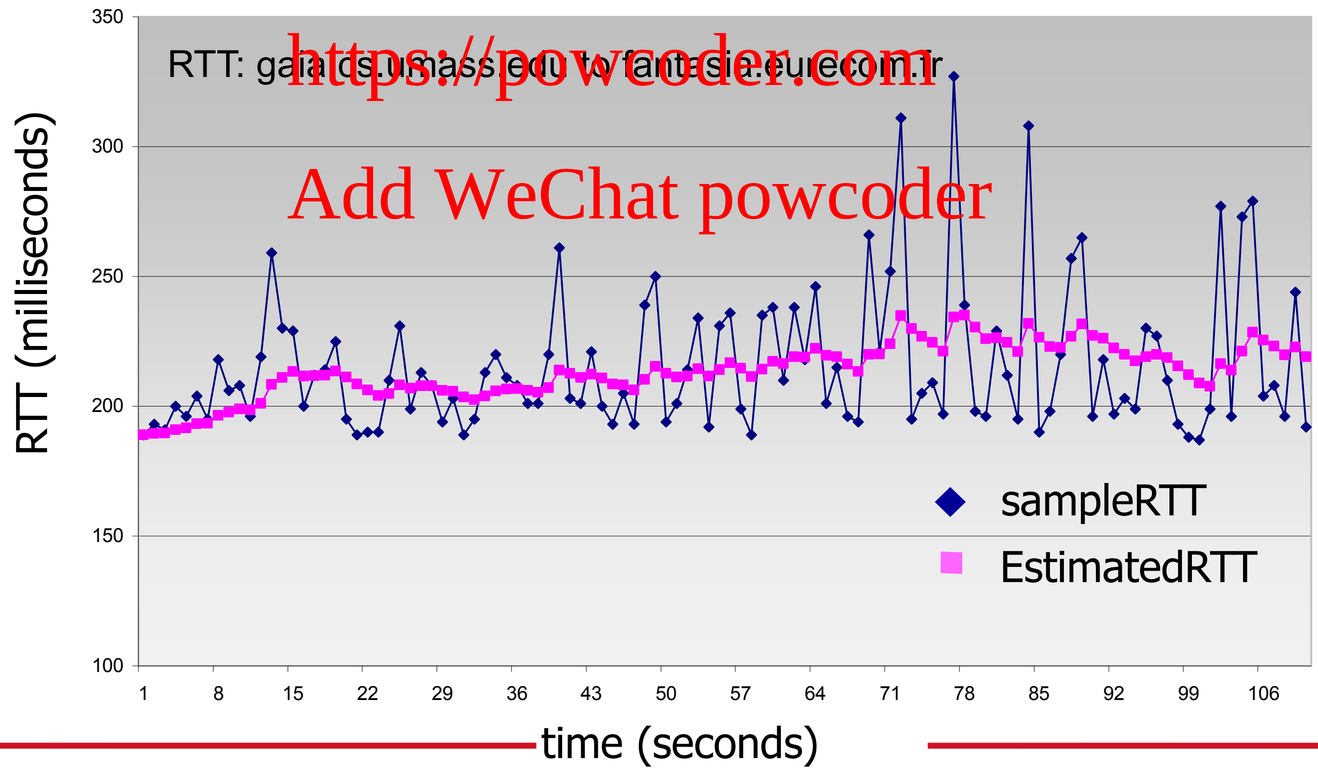


# TCP round trip time, timeout

**EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT**

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125





# TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:



```
DevRTT = (1-β)*DevRTT +
                β*|SampleRTT-EstimatedRTT|

                (typically, β = 0.25)
```



```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT

"safety margin"

# Reliable Data Transfer in TCP

# TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks



let's initially consider simplified TCP sender:
- ignore duplicate acks
- ignore flow control, congestion control

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# TCP sender events

*data rcvd from app:*

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unacked segment
  - expiration interval: `TimeOutInterval`

*timeout:*

- retransmit segment that caused timeout
- restart timer

*ack rcvd:*

- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments

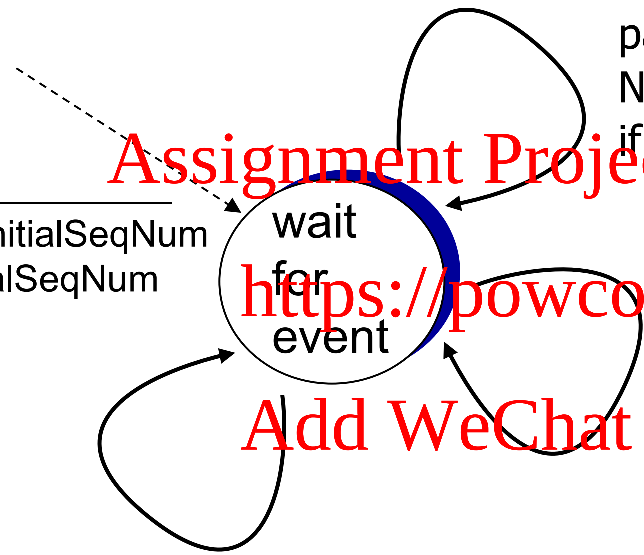
# TCP sender (simplified)

data received from application above

create segment, seq. #: NextSeqNum
pass segment to IP (i.e., “send”)
NextSeqNum = NextSeqNum + length(data)
if (timer currently not running)
start timer

$\Lambda$

NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum



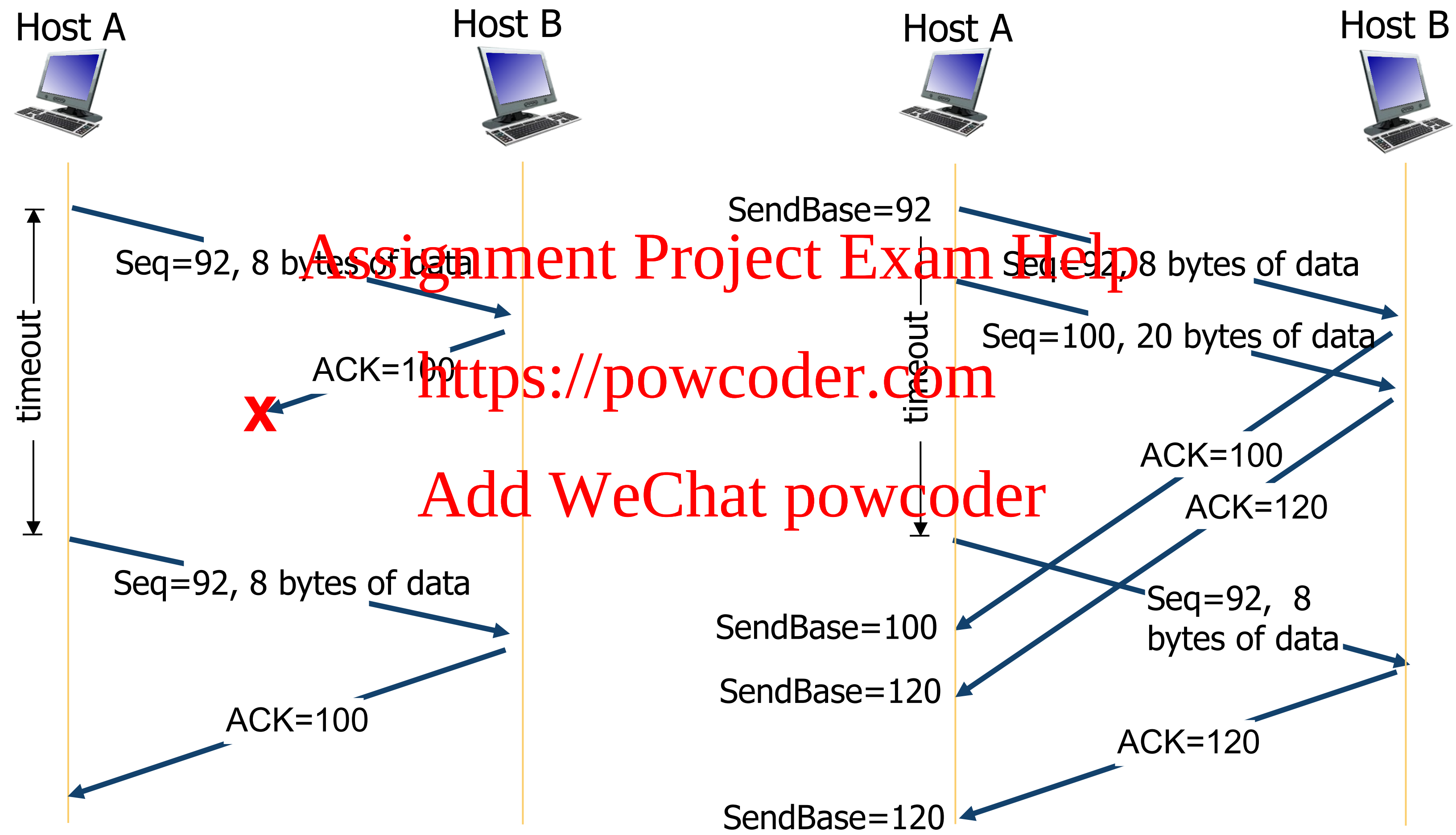

timeout

retransmit not-yet-acked segment
with smallest seq. #
start timer

ACK received, with ACK field value y

```
if (y > SendBase) {
   SendBase = y
   /* SendBase–1: last cumulatively ACKed byte */
   if (there are currently not-yet-acked segments)
        start timer
     else stop timer
  }
```

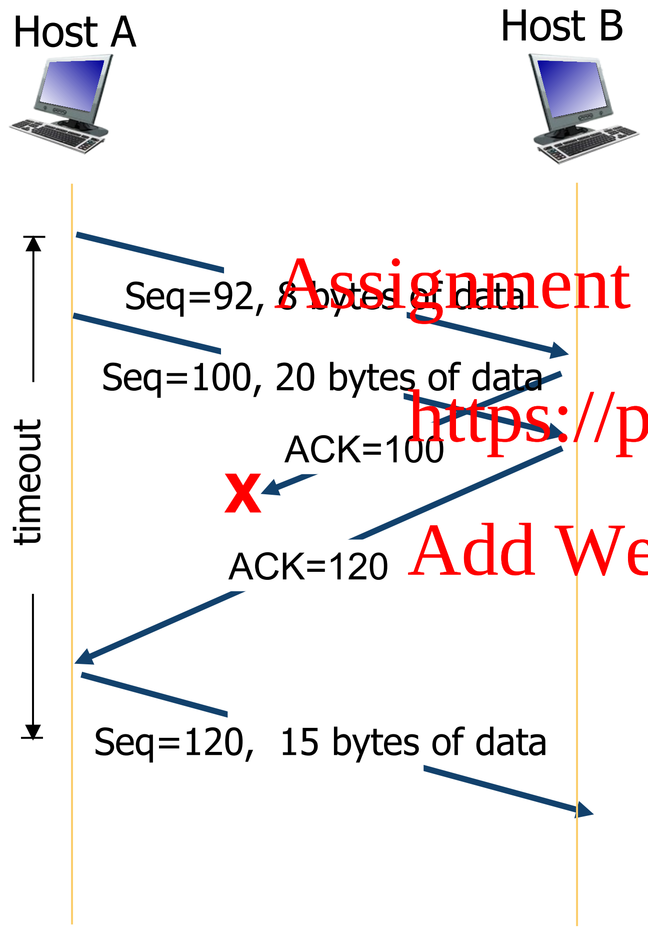
# TCP: retransmission scenarios




lost ACK scenario

premature timeout

# TCP: retransmission scenarios

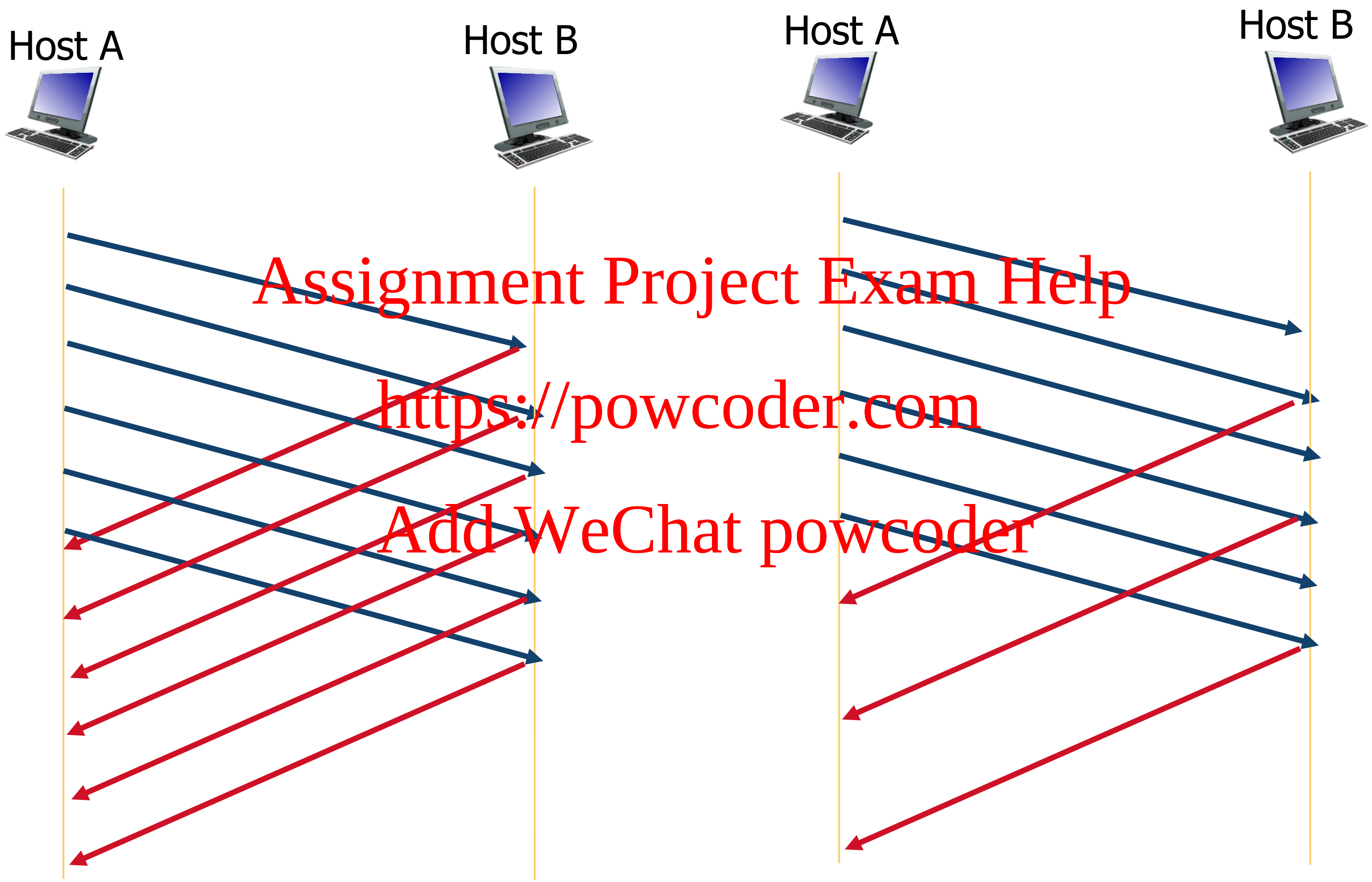



cumulative ACK

# TCP ACK generation [RFC 1122, RFC 2581]

| *event at receiver* | *TCP receiver action* |
|---|---|
| arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed | delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK |
| arrival of in-order segment with expected seq #. One other segment has ACK pending | immediately send single cumulative ACK, ACKing both in-order segments |
| arrival of out-of-order segment higher-than-expect seq. # . Gap detected | immediately send *duplicate ACK,* indicating seq. # of next expected byte |
| arrival of segment that partially or completely fills gap | immediate send ACK, provided that segment starts at lower end of gap |

# TCP ACK




Too many ACKs

cumulative ACK

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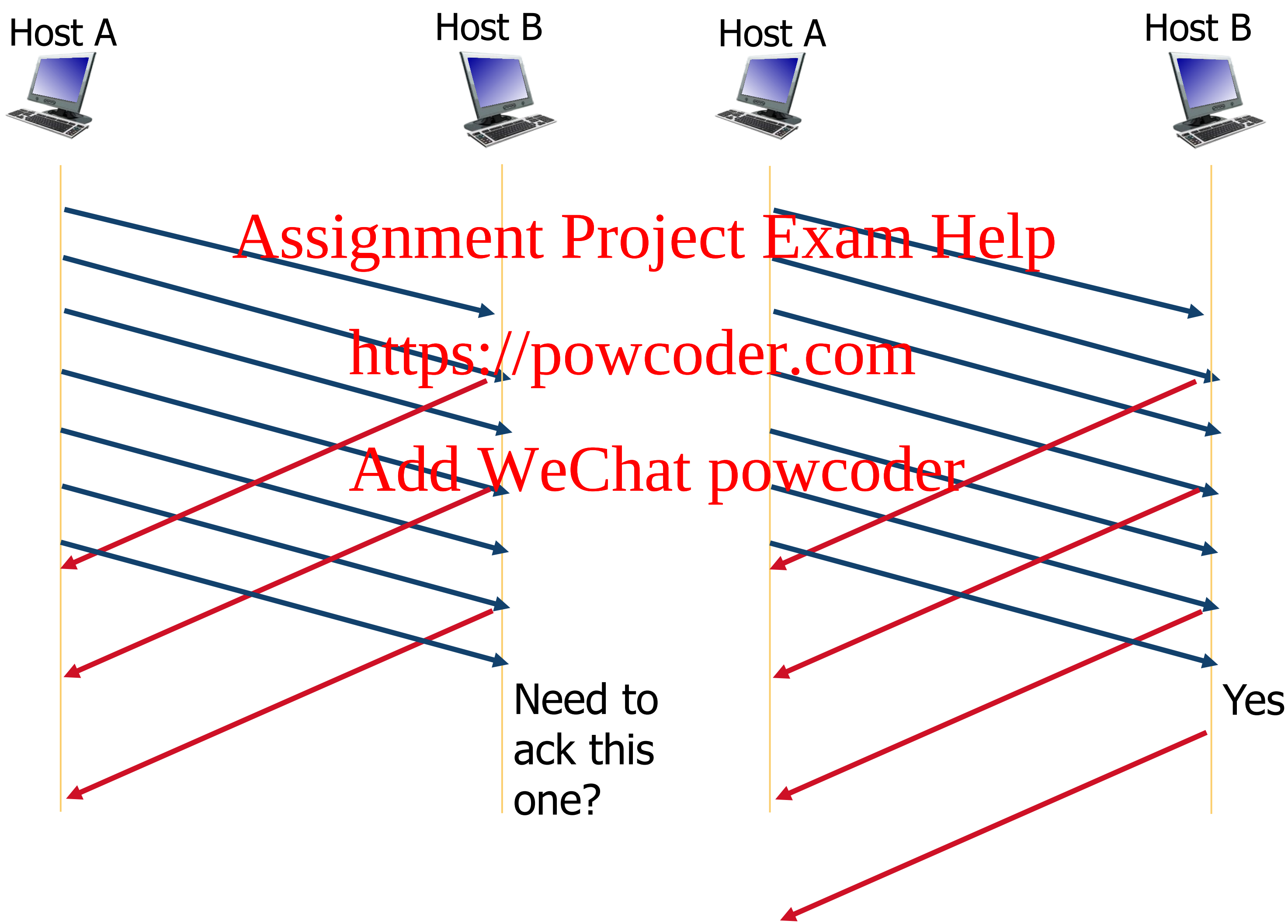
# TCP ACK



# TCP fast retransmit

- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.



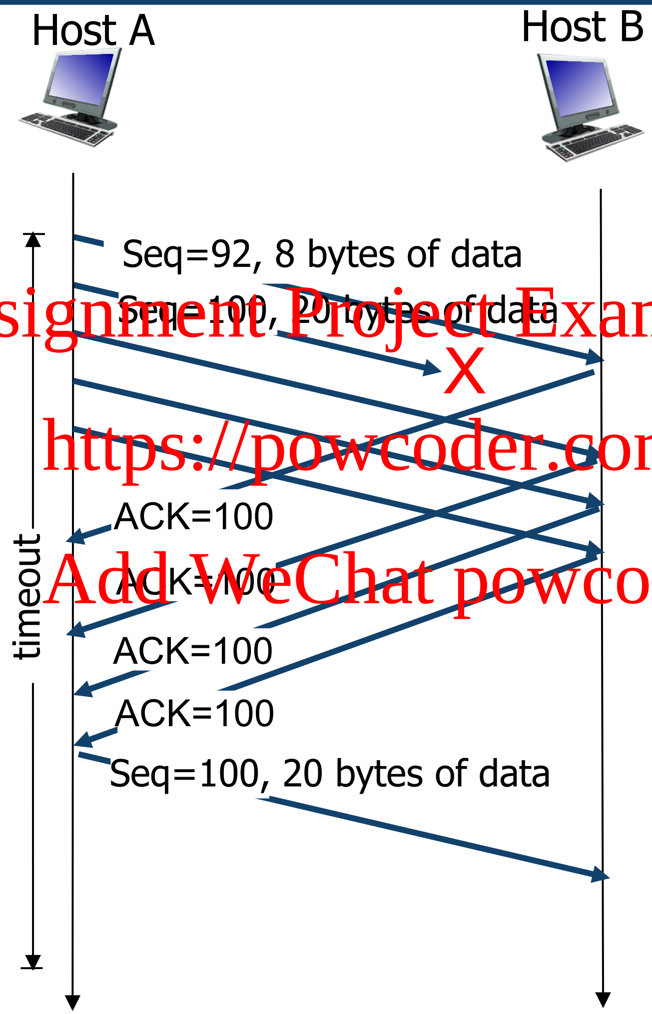
*TCP fast retransmit*

if sender receives 3 duplicate ACKs for same data

("triple duplicate ACKs"), resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout

# TCP fast retransmit






fast retransmit after sender
receipt of triple duplicate ACK

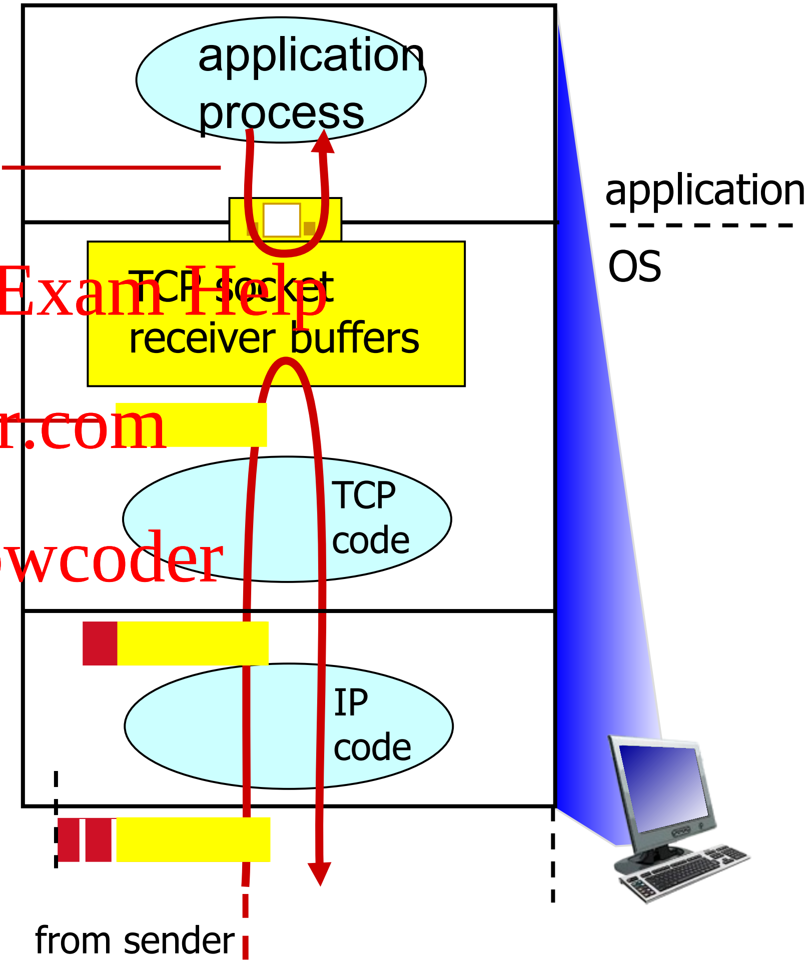
# Flow Control in TCP

# TCP flow control

application may remove data from TCP socket buffers ….

… slower than TCP receiver is delivering (sender is sending)





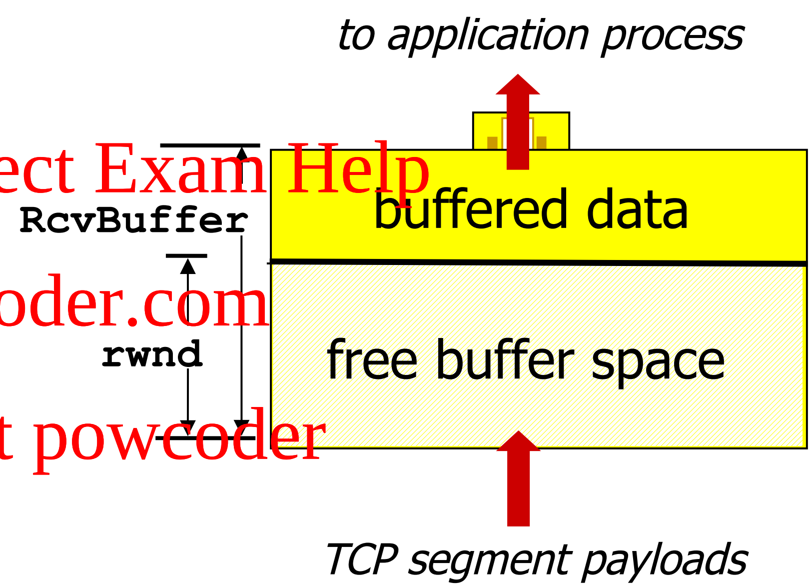

receiver protocol stack

***flow control***

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

# TCP flow control

- receiver "advertises" free buffer space by including **`rwnd`** value in TCP header of receiver-to-sender segments
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow




*receiver-side buffering*

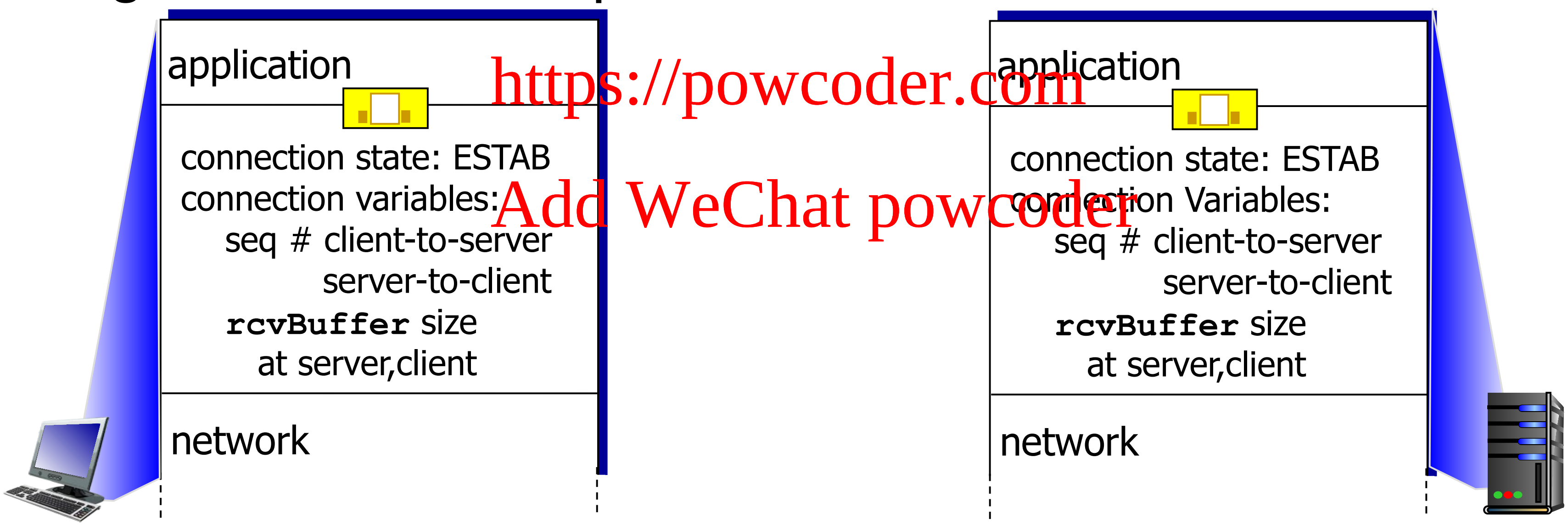
# Connection Management in TCP

# Connection Management

before exchanging data, sender/receiver "handshake":

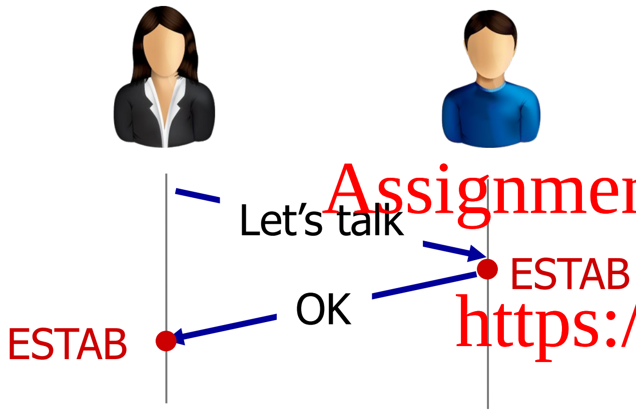
- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters





# Agreeing to establish a connection

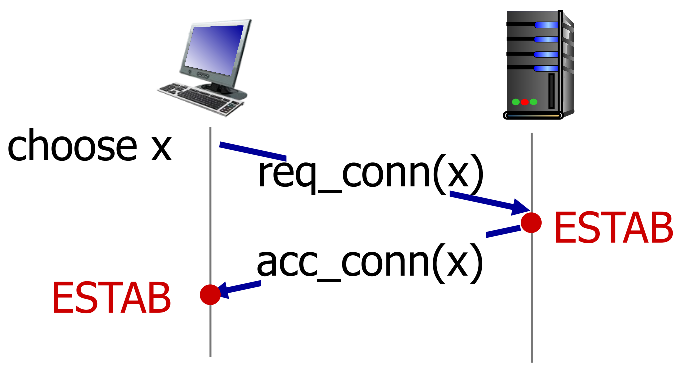
## 2-way handshake:




*Q:* will 2-way handshake always work in network?

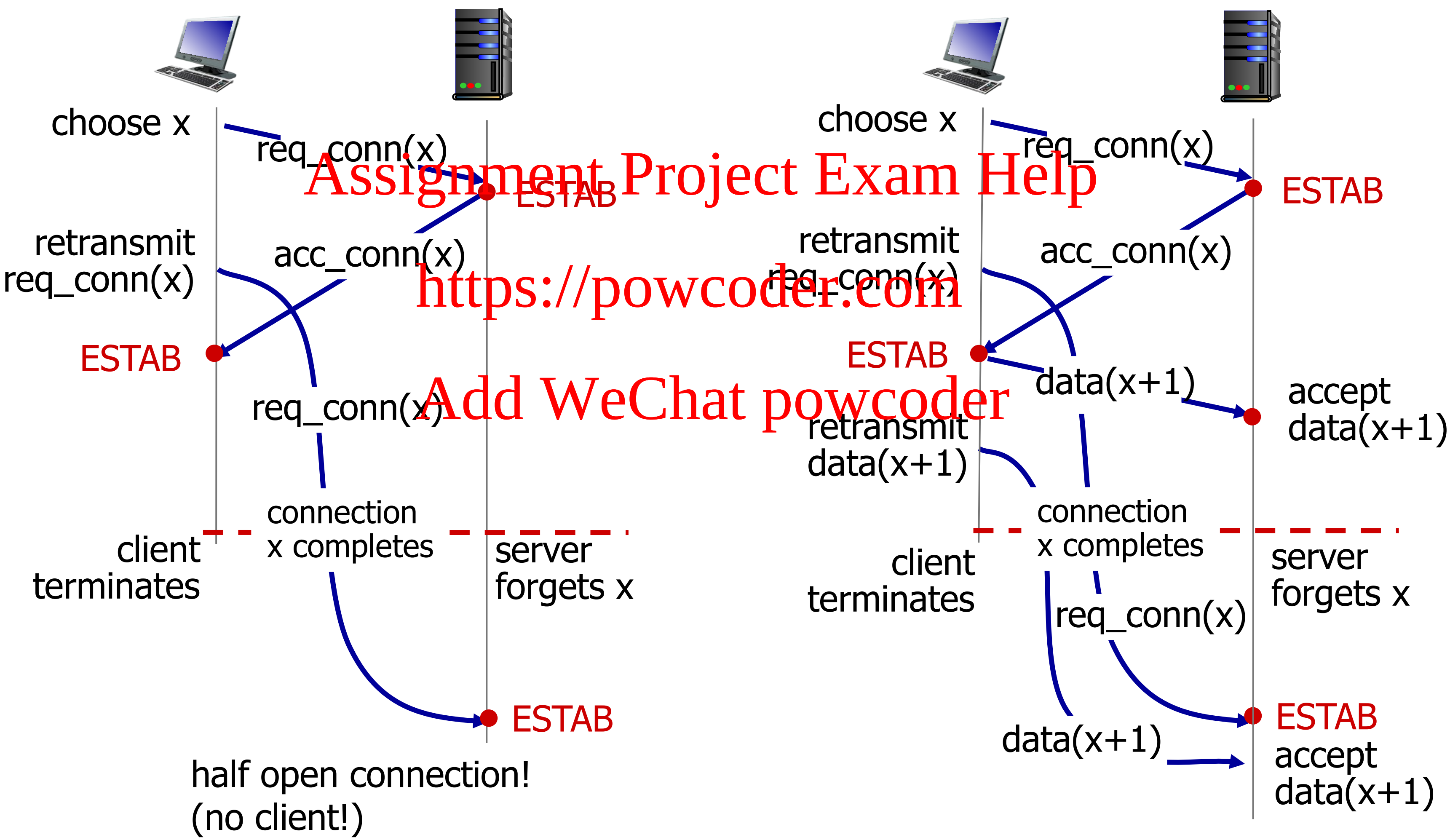
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side



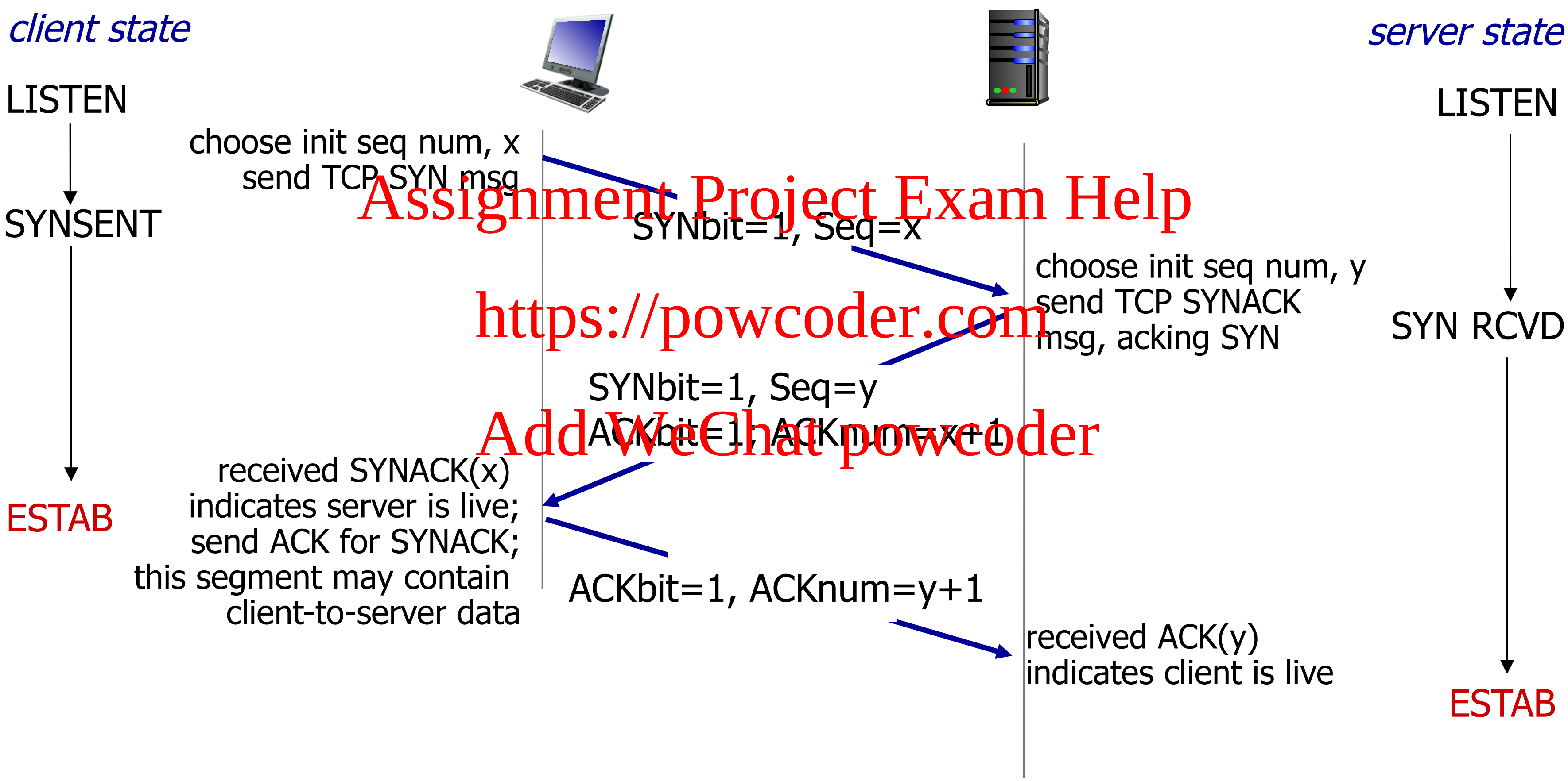


# Agreeing to establish a connection

## 2-way handshake failure scenarios:



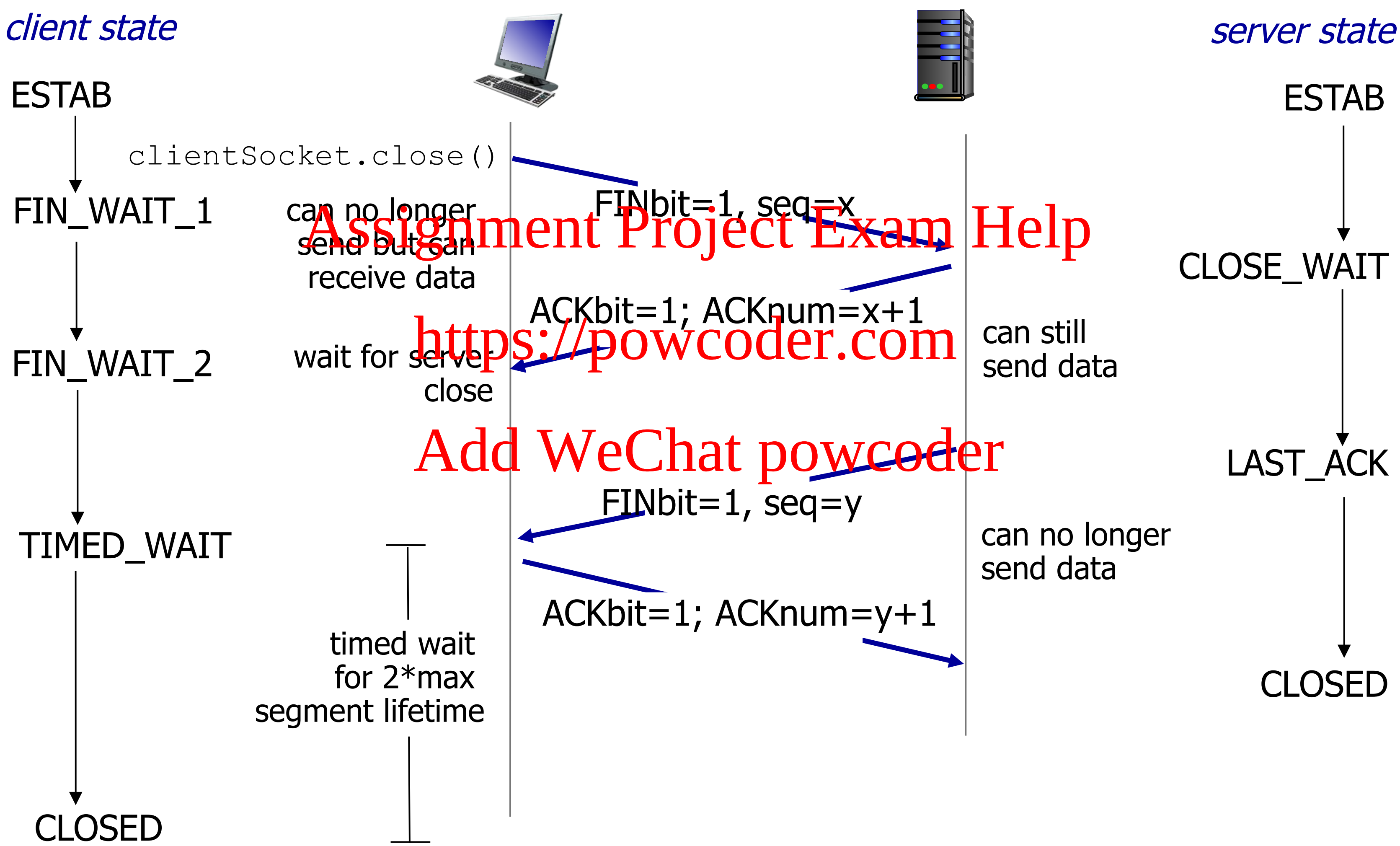
# TCP 3-way handshake



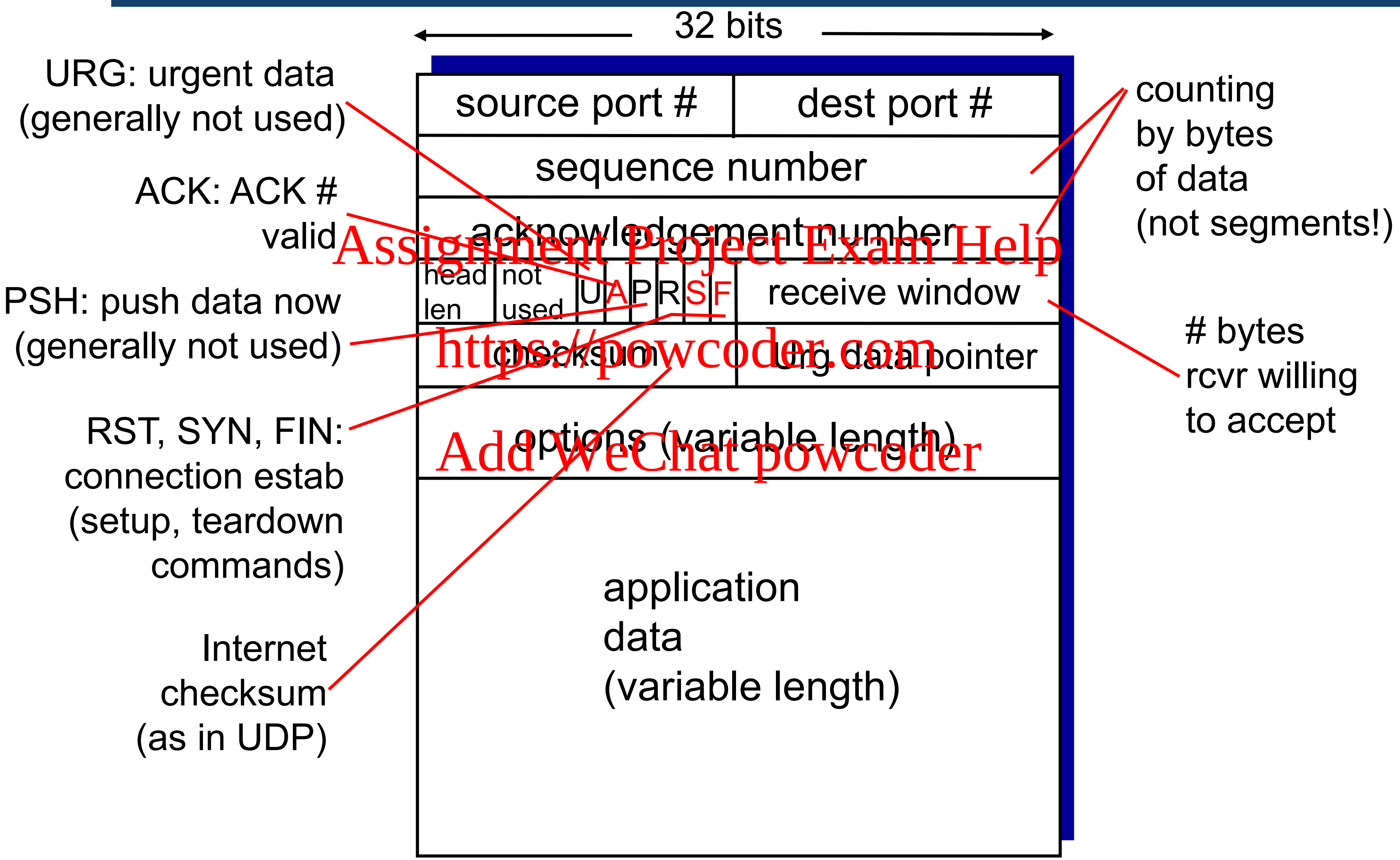
# TCP: closing a connection

› client, server each closes their side of connection
  - send TCP segment with FIN bit = 1
› respond to received FIN with ACK

# TCP: closing a connection



# TCP segment structure



# Principles of Congestion Control

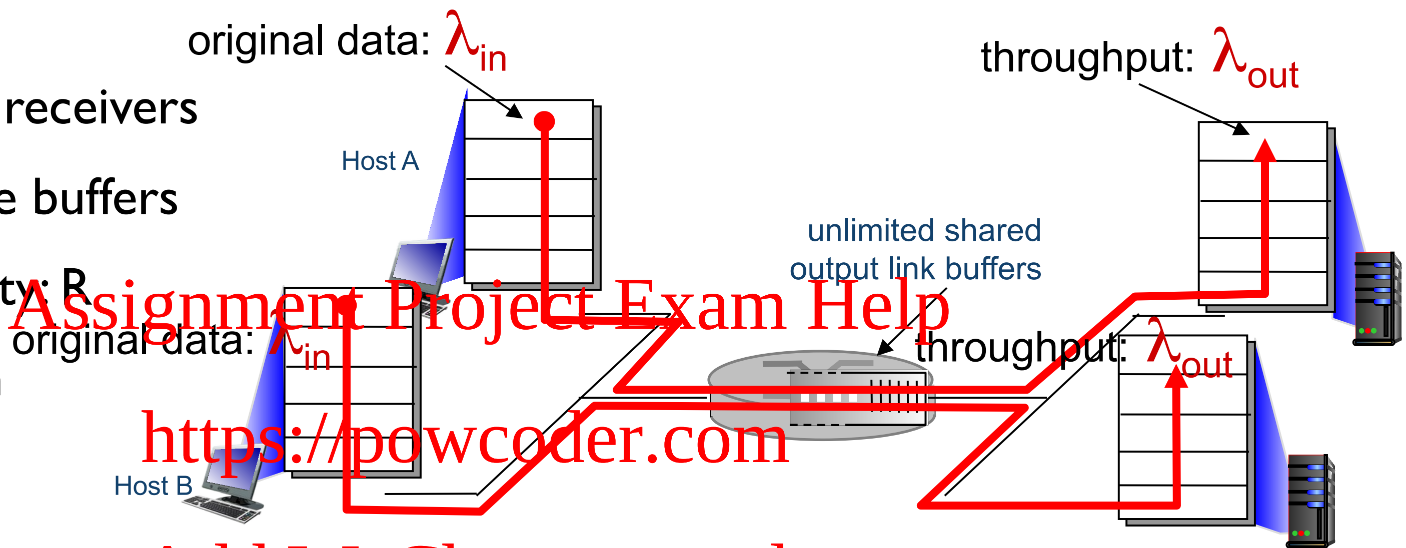
# Principles of congestion control

*congestion:*

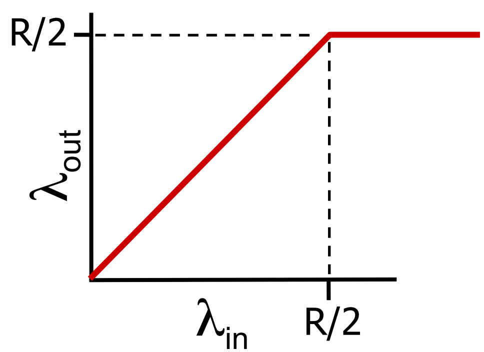
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

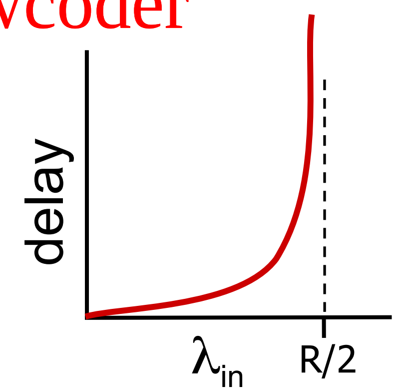
# Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission





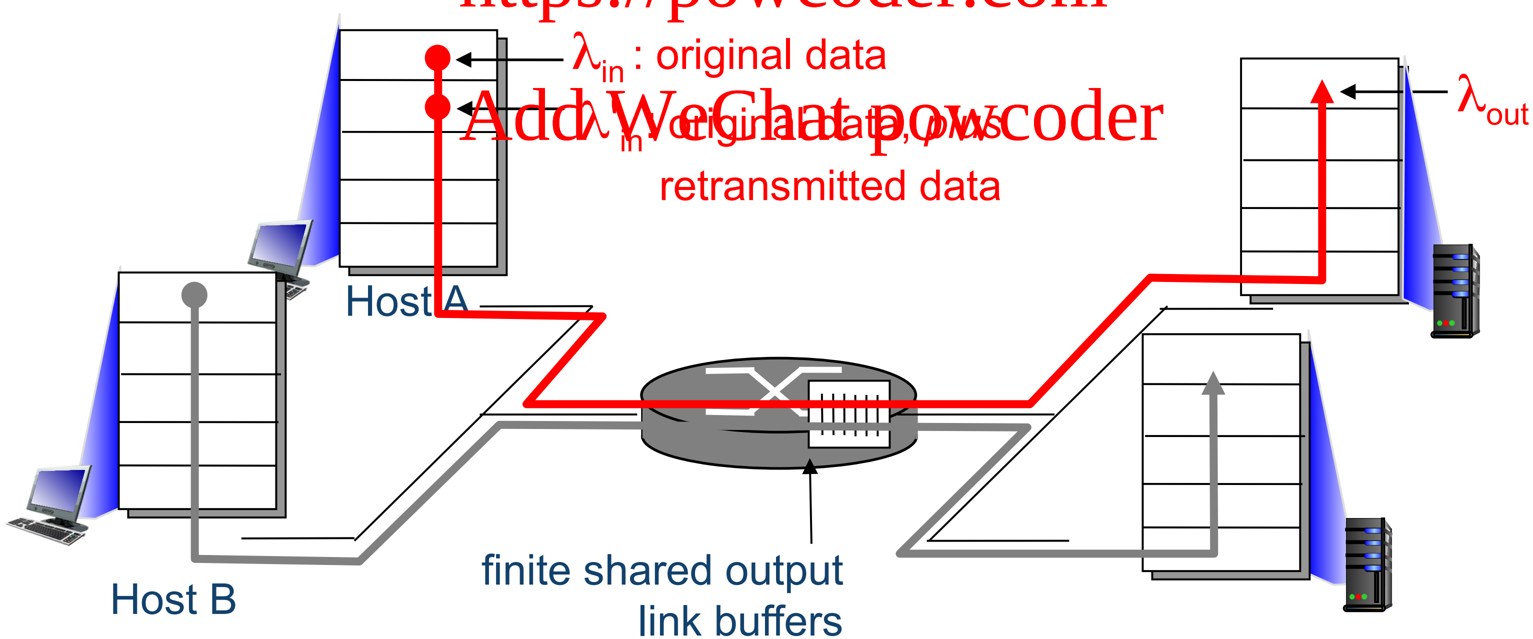

- maximum per-connection throughput: R/2

- large delays as arrival rate, $\lambda_{in}$, approaches capacity

# Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$、
    - Goodput
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

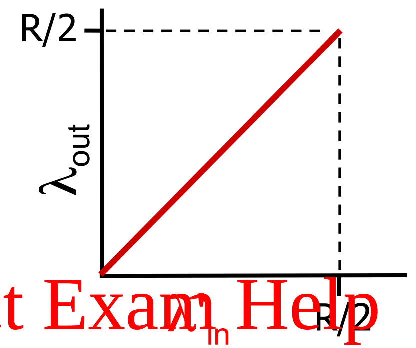




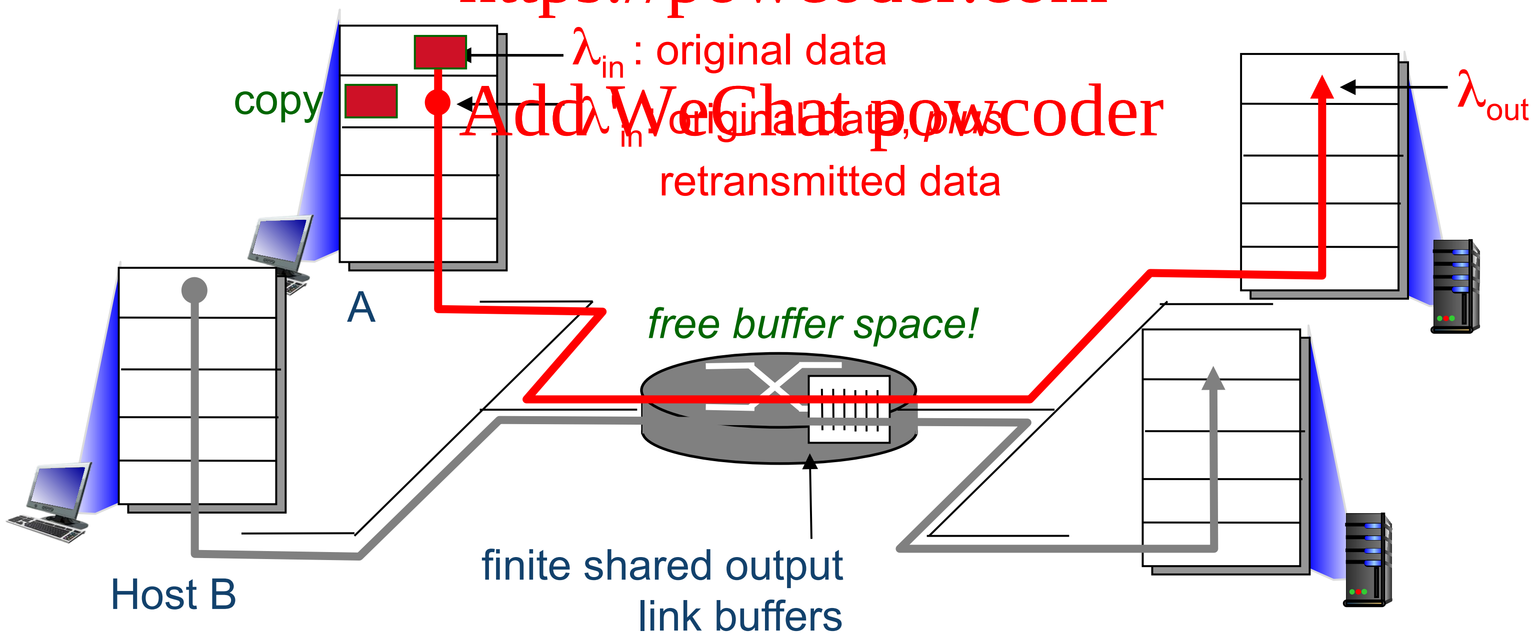
# Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available





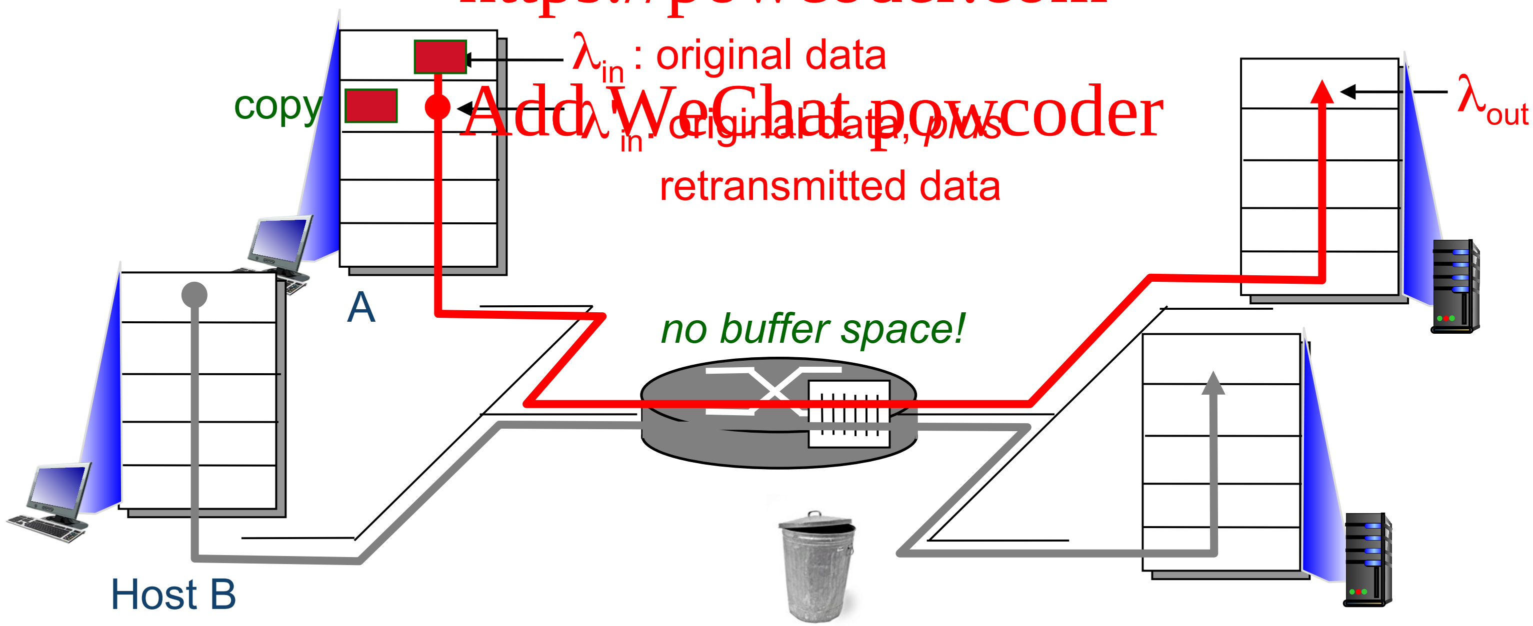
# Causes/costs of congestion: scenario 2

*Idealization: known loss*

packets can be lost, dropped at router due  to full buffers

- sender only resends if packet *known* to be lost





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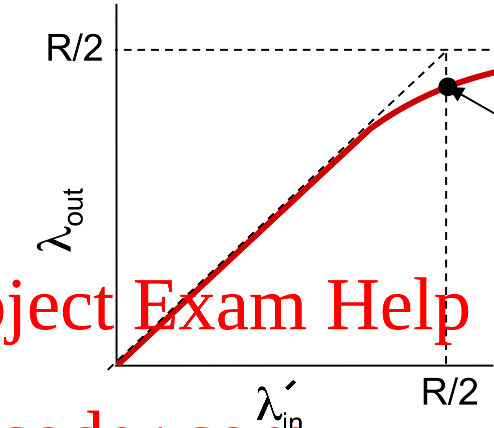
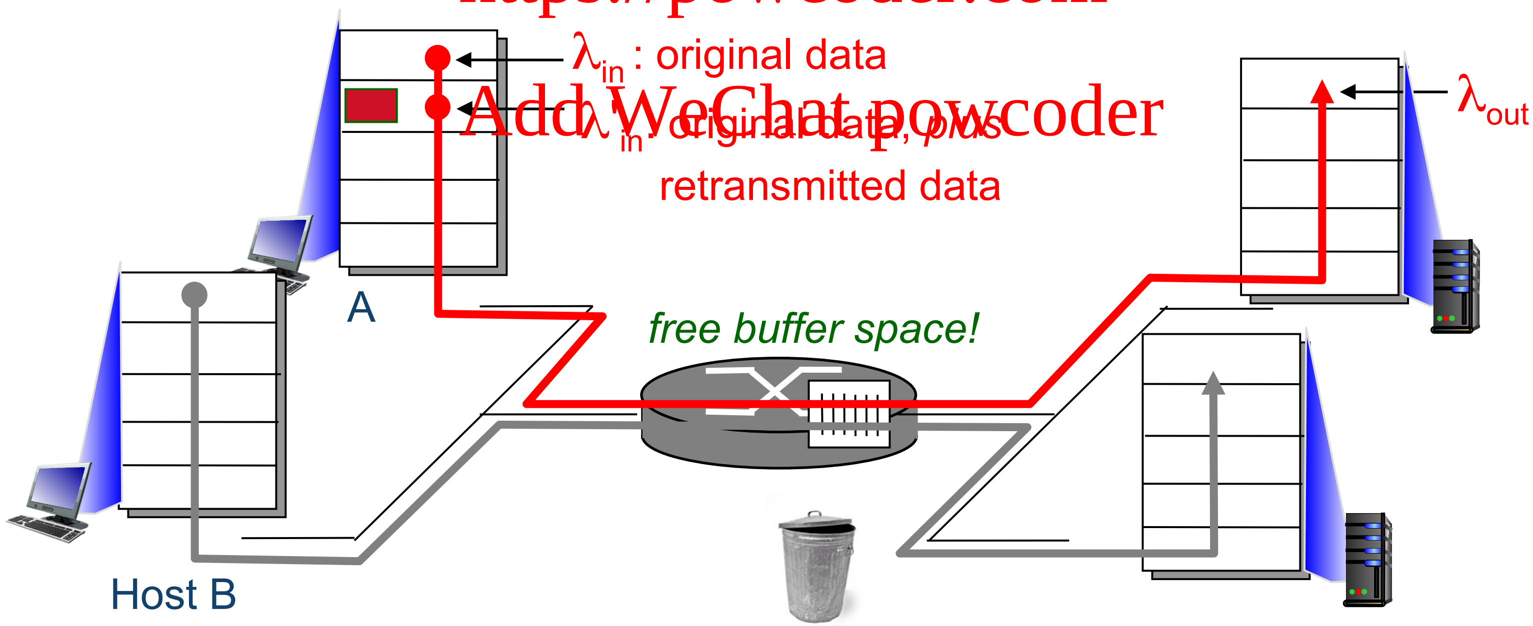



when sending at R/2, some packets are retransmissions but asymptotic goodput is still R/2

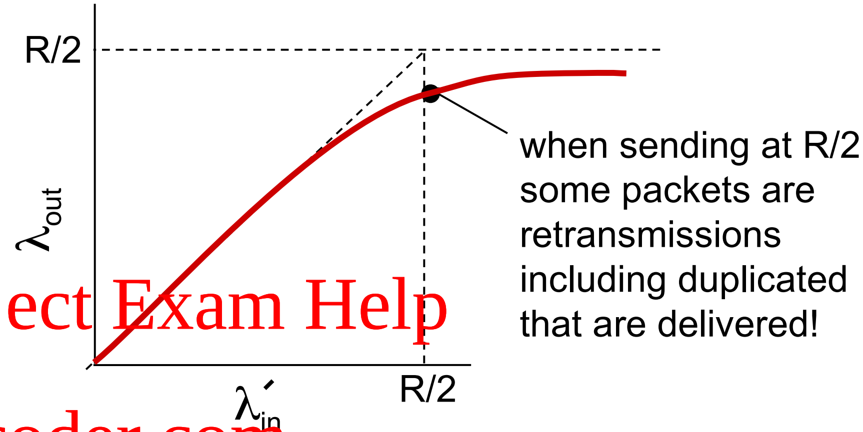




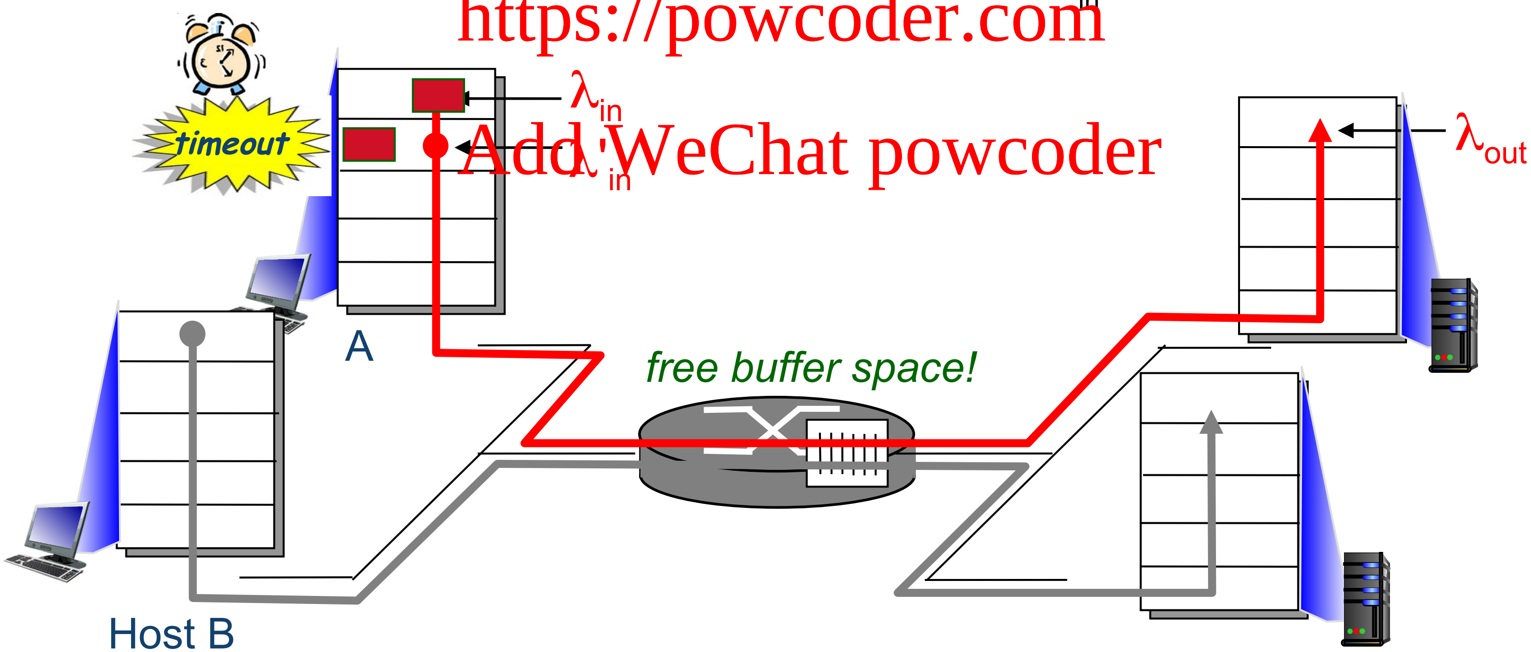
# Causes/costs of congestion: scenario 2

## *Realistic: duplicates*

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



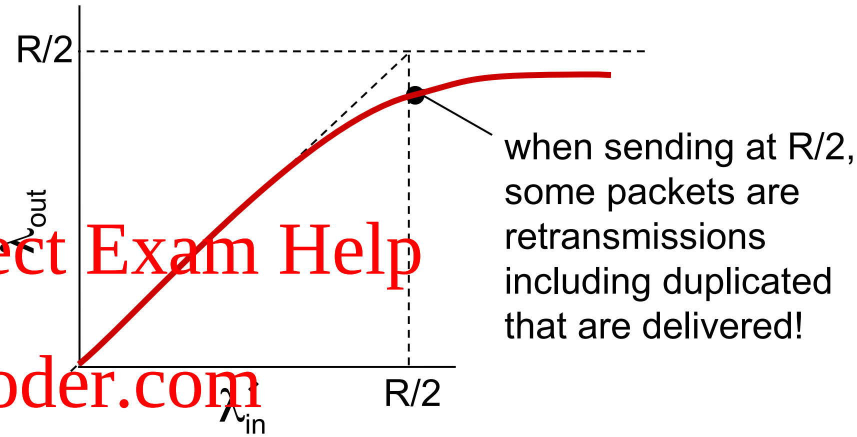


# Causes/costs of congestion: scenario 2

## *Realistic: duplicates*

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered






## "costs" of congestion:

- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

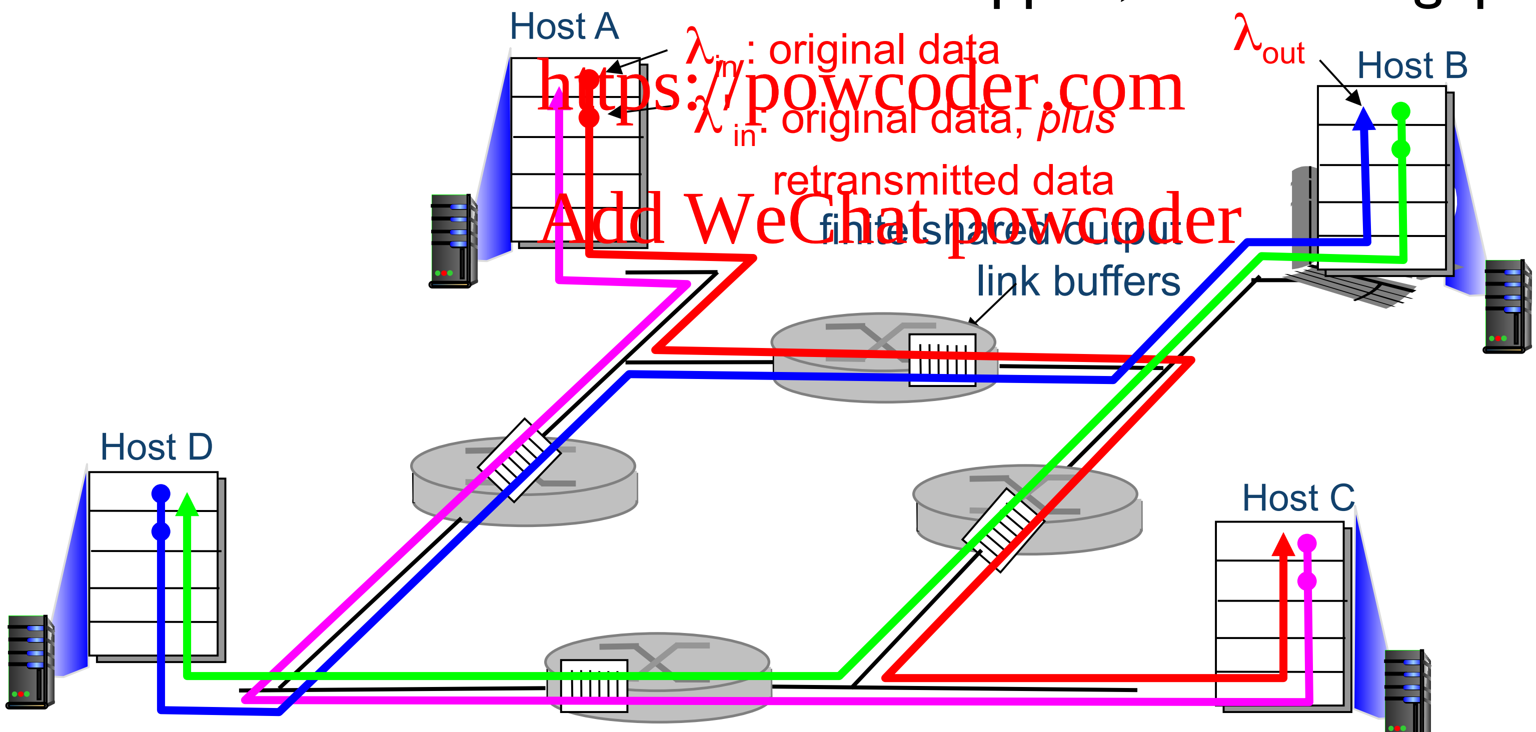
# Causes/costs of congestion: scenario 3

- four senders
- multihop paths
- timeout/retransmit

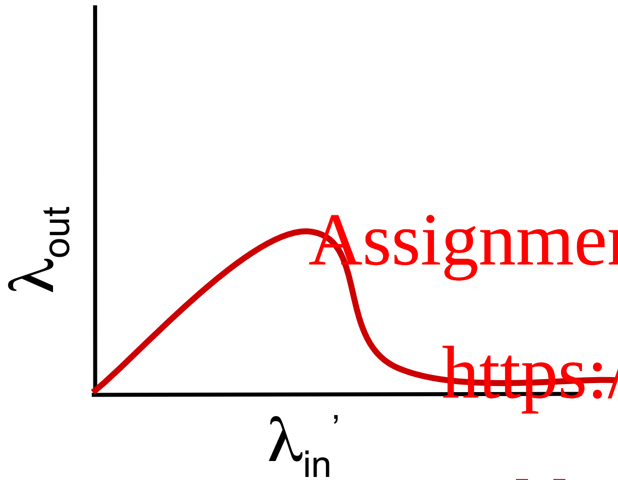
Q: what happens as $\lambda_{in}'$ increases ?

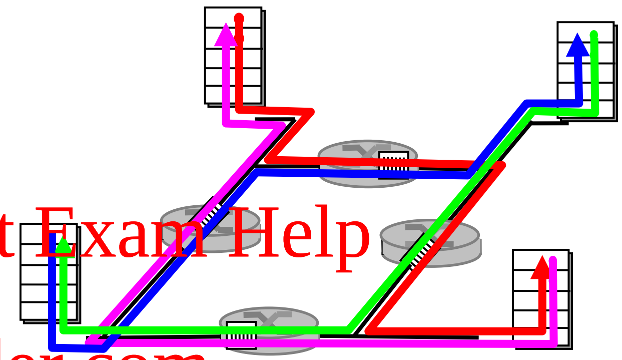
A: as red $\lambda_{in}'$ increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow$ 0





# Causes/costs of congestion: scenario 3





another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!

# Approaches towards congestion control

two broad approaches towards congestion control:







**end-end congestion control:**

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

**network-assisted congestion control:**

- routers provide feedback to end systems
  - single bit indicating congestion
  - explicit rate for sender to send at

# TCP Congestion Control

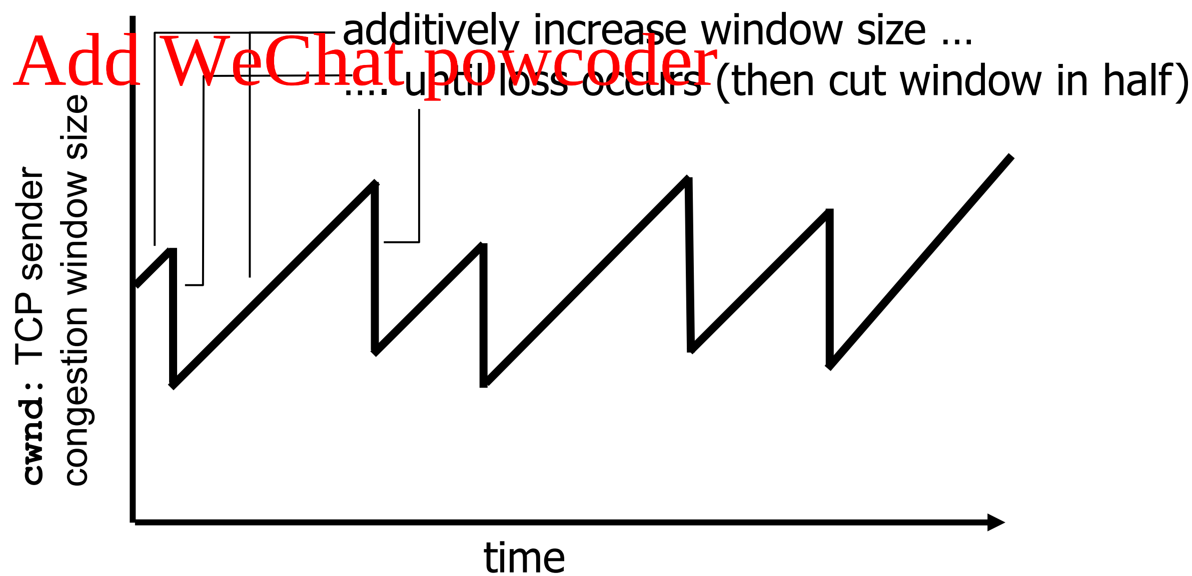
# TCP congestion control

## Additive increase multiplicative decrease (AIMD)

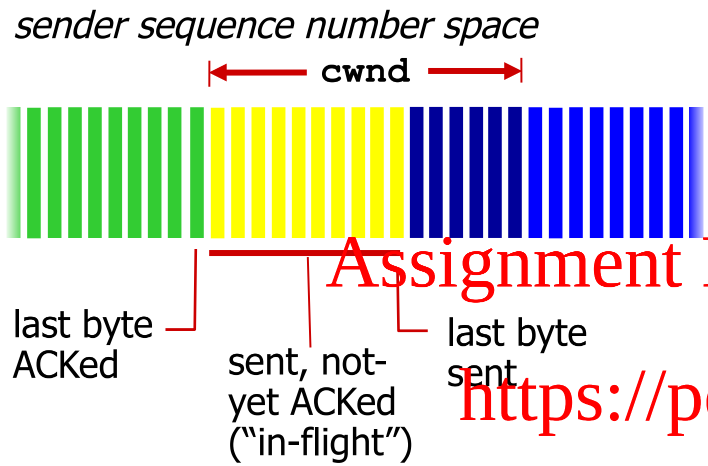
- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **cwnd** by 1 MSS (maximum segment size) every RTT until loss detected
  - *multiplicative decrease*: cut **cwnd** in half after loss



AIMD saw tooth behavior: probing for bandwidth



# TCP Congestion Control: details




› sender limits transmission:

`LastByteSent- LastByteAcked ≤ cwnd`

› `cwnd` is dynamic, function of perceived network congestion

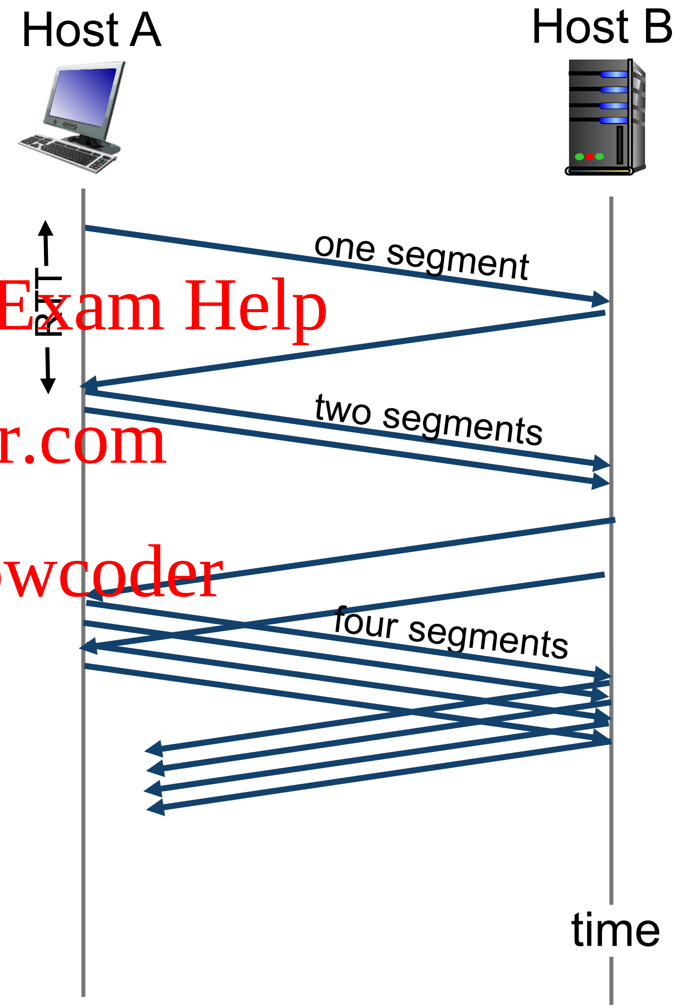
## *TCP sending rate:*

› *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\text{rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

# TCP Slow Start

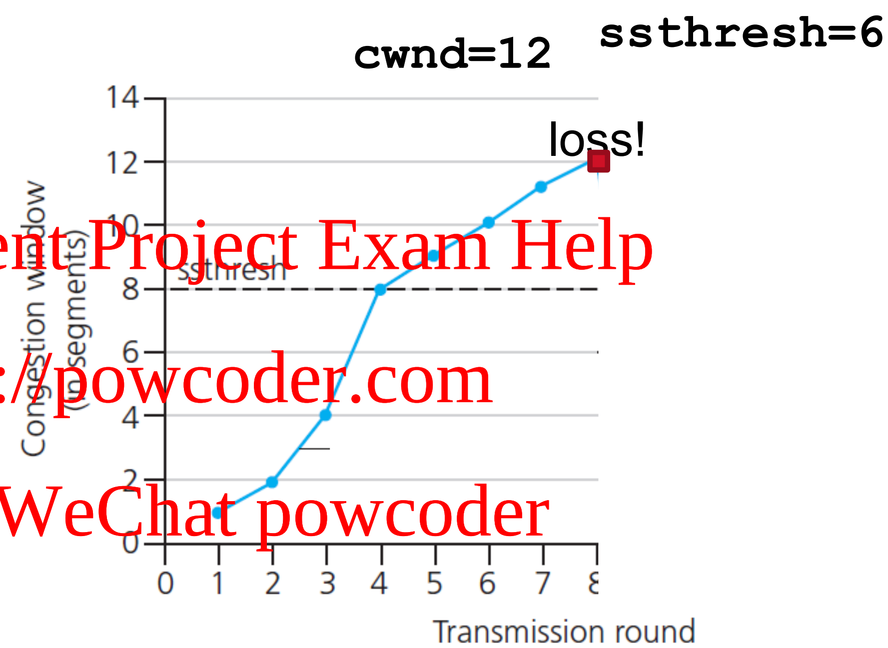
- when connection begins, increase rate exponentially:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow but ramps up exponentially fast
- when should the exponential increase switch to linear (additive increase)?



# TCP: switching from slow start to CA

**Q:** when should the exponential increase switch to linear?

**A:** cwnd reaches **`ssthresh`**






Implementation:

- At beginning **`ssthresh`**, specified in different versions of TCP
- (In this example **`ssthresh=8 segment`**)
- on loss event, **`ssthresh`** is set to 1/2 of **`cwnd`** just before loss event

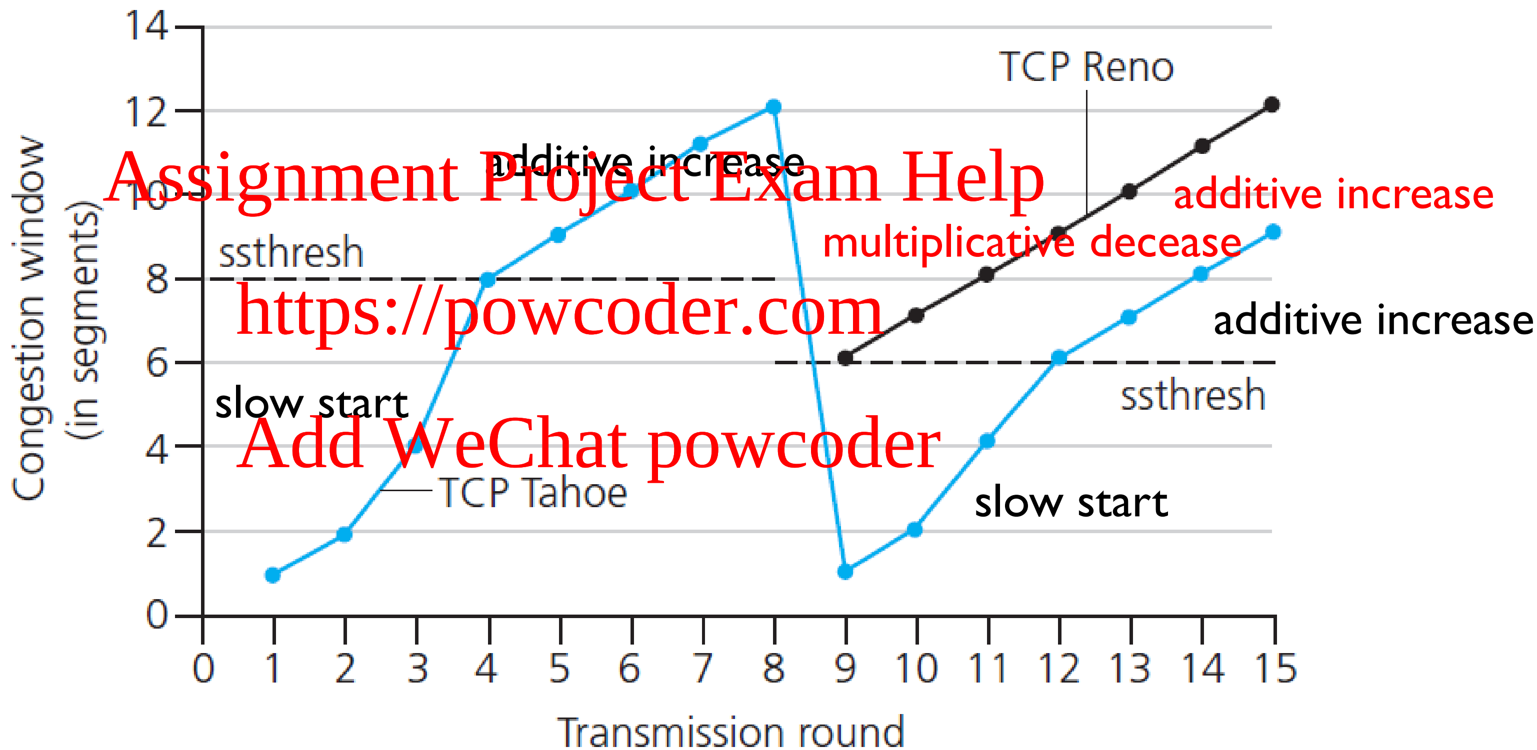
# TCP: detecting, reacting to loss

- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to **`ssthresh`**, then grows linearly



- loss indicated by 3 duplicate ACKs:
  - TCP Tahoe, same as loss indicated by timeout, always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)
  - TCP RENO
    - **`cwnd`** is cut in half window then grows linearly (additive increase)
    - fast recovery

# TCP: switching from slow start to CA



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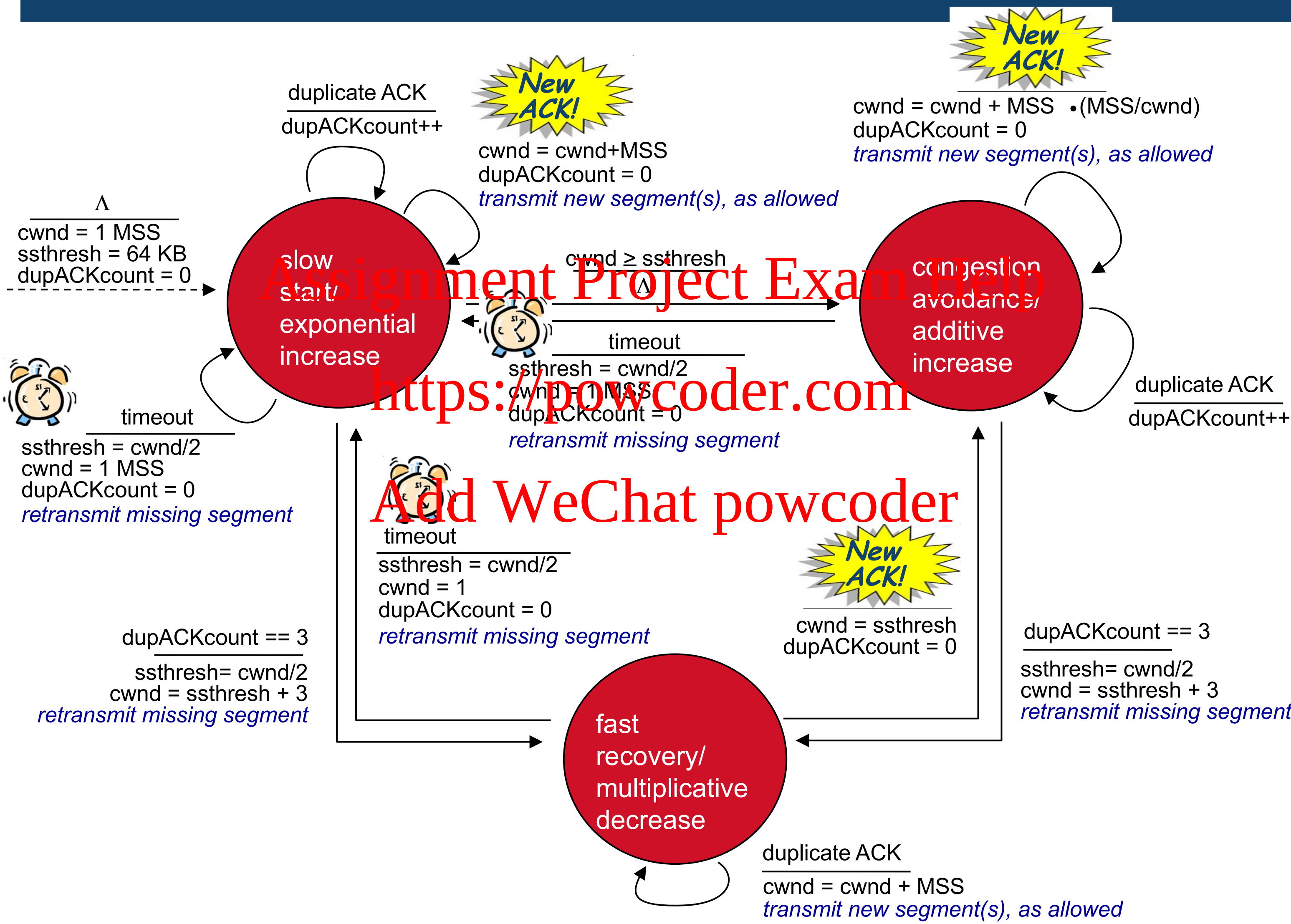
# Summary: TCP Reno Congestion Control

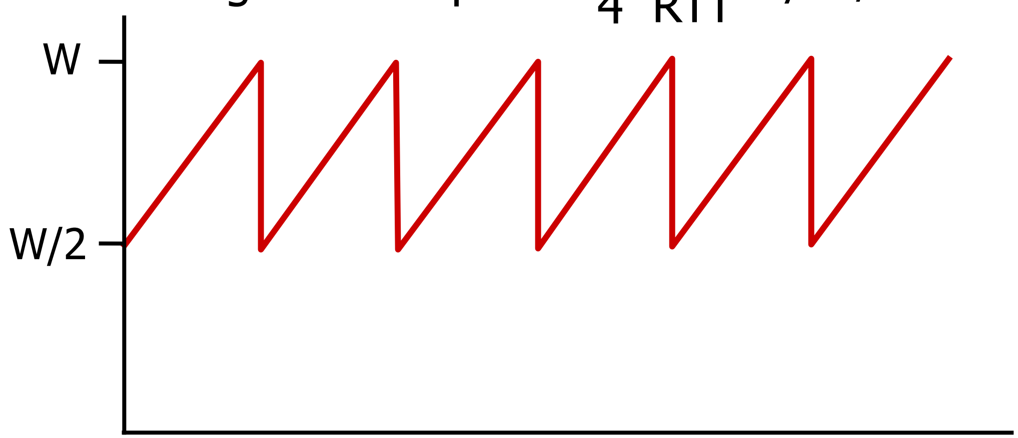


# TCP Reno throughput

- avg. TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT



$$\text{avg TCP thruput} = \frac{3}{4}\frac{W}{RTT} \text{ bytes/sec}$$



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# TCP Futures: TCP over “long, fat pipes”

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments

- throughput in terms of segment loss probability, L [Mathis 1997]:


$$\text{TCP throughput} = \frac{1.22 \cdot MSS}{RTT \sqrt{L}}$$



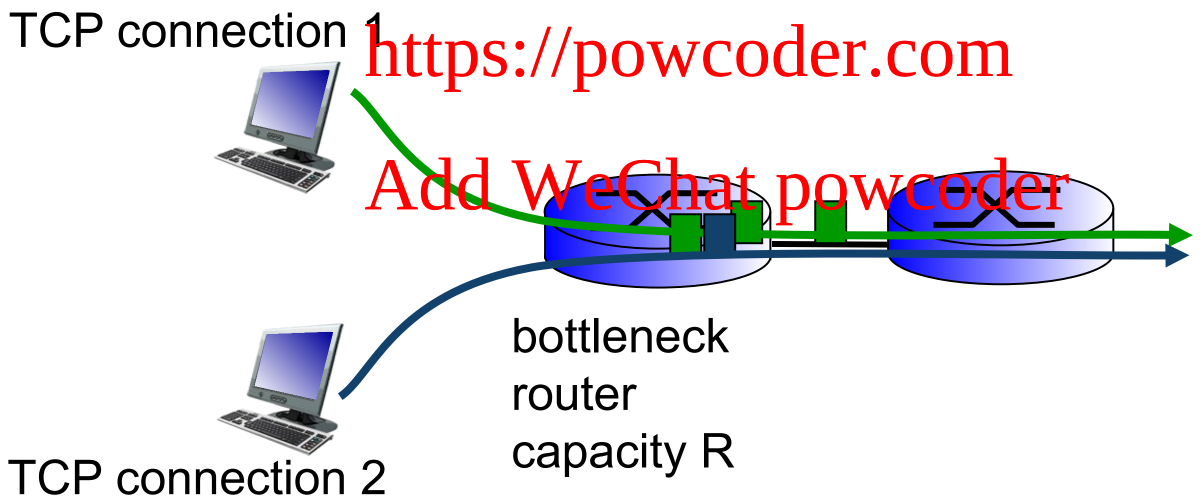
➜ to achieve 10 Gbps throughput, need a loss rate of $L = 2 \cdot 10^{-10}$ – *a very small loss rate!*

- new versions of TCP for high-speed
  - Vegas, Westwood, CUBIC, etc.

# TCP Fairness

*Fairness:* K TCP sessions share same bottleneck link of bandwidth R, each has average rate of R/K



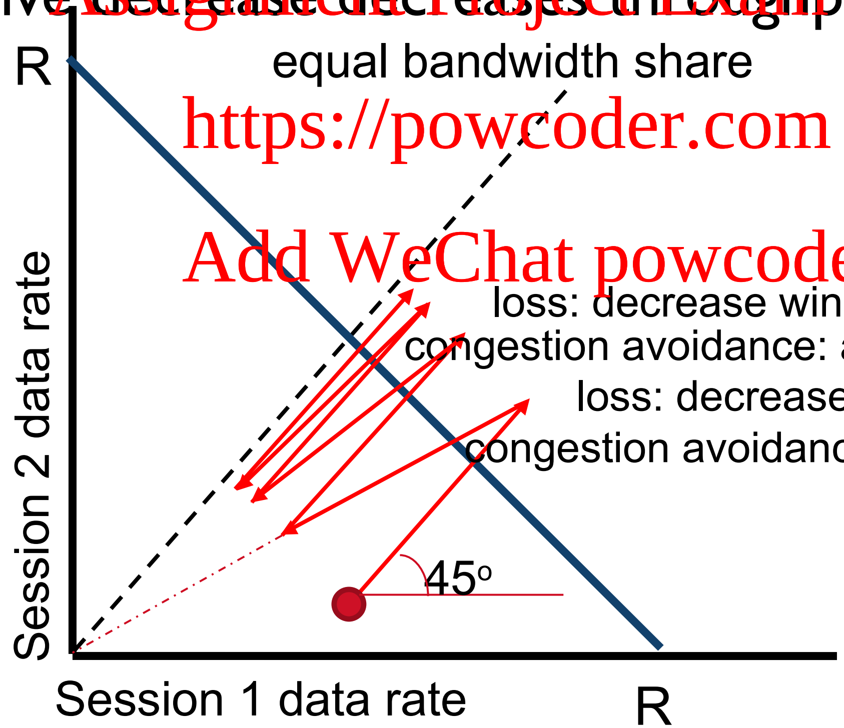


# Why is TCP fair?

two competing sessions:

- additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally





# Fairness (more)

### *Fairness and UDP*

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss

### *Fairness, parallel TCP connections*

- application can open multiple parallel connections between two hosts
- e.g. link of rate R
  - App 1 asks for 1 TCP, gets 0.1R
  - App 2 asks for 9 TCPs, gets 0.9R

# Final word

*Window size = min (rwnd, cwnd)*



*receive window* *congestion window*



*flow control* *congestion control*

# Conclusion

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - connection setup, teardown
  - flow control
  - congestion control
- instantiation, implementation in the Internet
  - UDP
  - TCP